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(54) Method for electronically enlarging the distance between two acoustical/electrical transducers and hearing aid apparatus

(57) For beam forming acoustical signals the phase difference of the output signals of two acoustical/electrical transducers is determined (27) and is multiplied by a factor (30). One of the two output signals of the at least two transducers is phase shifted by an amount according to the multiplication result. This phase shifted signal and the signal of the second transducer are led to a signal processing unit, wherein beam forming on these at least two signals is performed. Thereby, it becomes possible to perform beam forming as if the transducers were mutually distant by more than they physically are.

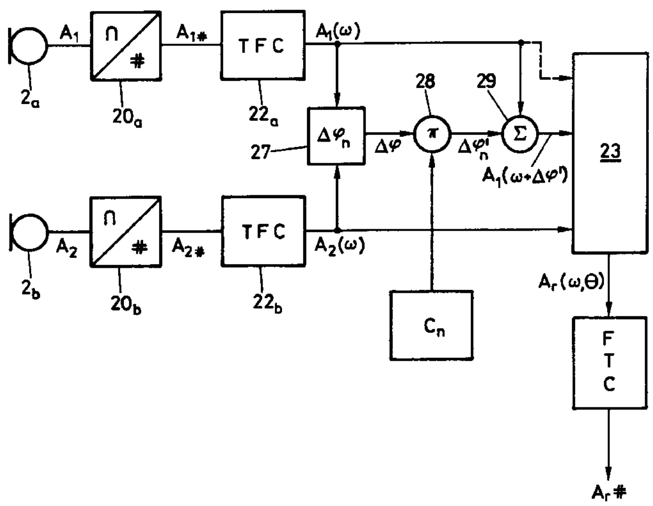

FIG.7

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Description

The present invention is generically directed on a technique according to which acoustical signals are received by at least two acoustical/electrical transducers as e.g. by multidirectional microphones, respective output signals of such transducers are electronically computed so as to generate a result signal which represents said acoustical signals weighted by a spatial characteristic of amplification as a function of spatial angle under which the acoustical signal impinges on the two transducers provided. Thus, the result signal represents the received acoustical signal weighted by a spatial amplification characteristic as if reception of the acoustical signals had been done by means of e.g. an antenna with an according reception lobe or beam.

Figure 1 most generically shows such known technique for such "beam forming" on acoustical signals. Thereby, at least two multidirectional acoustical/electrical transducers 2_a and 2_b are provided, which both convert acoustical signal irrespective of their impinging direction θ and thus substantially unweighted with respect to impinging direction θ into electrical output signals A_1 and A_2. The output signals A_1 and A_2 are fed to an electronic signal processor unit 3 which generates from the input signals A_1, A_2 a result signal A_r. As shown within the block of unit 3 in that unit 3 the signals $A_{1,2}$ are treated to result in the result signal A_r which represents either of A_1 or A_2, but additionally weighted by the spatial amplification function $F(\theta)$. Thus, acoustic signals may selectively be amplified dependent from the fact under which spatial angle θ they impinge, i.e. under which spatial angle the transducer arrangement 2a, 2b "sees" an acoustical source.

One approach to perform signal processing within processor unit 3 shall be exemplified with the help of Fig. 2. Thereby, all such approaches are based on the fact that due to a predetermined mutual distance p_P of the two transducers 2_a and 2_b, there occurs a time-lag dt between reception of an acoustical signal at the transducers 2_a, 2_b.

Considering a single frequency - ω - acoustical signal, received by the transducer 2_a, this transducer will generate an output signal

$$A_1 = A \cdot \sin\omega t, \qquad (1)$$

whereas the second transducer 2_b will generate an output signal according to

$$A_2 = A \cdot \sin\omega(t+dt), \qquad (2)$$

whereat dt is given by

$$dt = \frac{p_P \sin\theta}{c} \qquad (3)$$

therein, c is the sound velocity.

By time-delaying A_1 by an amount

$$\tau = p_P/c \qquad (4)$$

and forming the result signal A_r from the difference of time-delayed signal A_1', namely from

$$A_1' = A \cdot \sin\omega(t+\tau), \text{ and} \qquad (5)$$

$$A_2 = A \cdot \sin\omega(t+dt), \qquad (2)$$

there results, considered at the frequency ω, a spatially cardoid weighted signal A_r as shown in the block of processing unit 3:

$$|A_r| = |A_1' - A_2| = 2A \sin(\omega(\tau-dt)/2) \qquad (6)$$
$$= 2A \sin(\omega(\tau-p_P{}^*\sin\theta/c)/2).$$

At $\theta = 90°$ A_r becomes zero and
at $\theta = -90°$ A_r becomes

$$A_{rmax} = 2A \sin \omega\, p_P/c. \qquad (7)$$

Such processing of the output signals of two omnidirectional order transducers leads to a first order cardoid weighing function $F_1(\theta)$ as shown in Fig. 3. By respectively selecting transducers with higher order acoustical to electrical conversion characteristic and/or by using more than two transducers, higher order - m - weighing functions $F_m(\theta)$ may be realised.

In Fig. 4 there is shown the amplitude A_{rmax}-characteristic, resulting from first order cardoid weighing as a function of frequency $f = \omega/2\pi$. Additionally, the respective function for a second order cardoid weighing function $F_2(\theta)$ is shown. Thereby, there is selected a distance p_P of the two transducers 2_a and 2_b of fig. 1 to be 12 mm.

As may clearly be seen at a frequency f_r which is

$$f_r = c/(4p) \tag{8}$$

maximum amplification occurs of +6 dB at the first order cardoid and of +12 dB at a second order cardoid. For $p_P = 12$ mm, f_r is about 7 kHz.

From fig. 4 a significant roll-off for low and high frequencies with respect to f_r is recognised, i.e. a significant decrease of amplification.

Techniques for such or similar type of beam forming are e.g. known from the US 4 333 170 - acoustical source detection -, from the European patent application 0 381 498 directional microphone - or from Norio Koike et al., "Verification of the Possibility of Separation of Sound Source Direction via a Pair of Pressure Microphones", Electronics and Communications in Japan, Part 3, Vol. 77, No. 5, 1994, page 68 to 75.

Irrespective of the prior art techniques used for such beam forming with at least two transducers, the distance p_P is an important entity as may be seen e.g. from formula (8).

Formula (8) may be of no special handicap if such technique is used for narrow band signal detection or if no serious limits are encountered for geometrically providing the at least two transducers at a large mutual distance p_P

Nevertheless, and especially for hearing aid applications, the fact that f_r is inversely proportional to the distance p_P of the transducers is a serious drawback, due to the fact that for hearing aid applications the audio frequency band up to about 4 kHz for speech recognition should be detectable by the at least two transducers which further should be mounted with the shortest possible mutual distance p_P.

It is thus a first object of the present invention to remedy the drawbacks encountered with respect to p_P dependency of acoustical "beam forming".

The first object of the present invention is reached by providing a method for electronically changing the distance between a first and a second acoustical/electrical transducer, generating, respectively, first and second electrical output signals which represent substantially simultaneously impinging acoustical signals, whereby the first and second electrical output signals are electronically treated so as to generate a result signal which result signal is a function of acoustical signals at least substantially simultaneously received at the two transducers and amplified by a spatial reception characteristic of amplification as a function of spatial angle under which the acoustical signals impinge on the two transducers, which method further comprises the steps of generating a third signal from at least one of the first and second output signals by shifting the phase of at least one of said output signals by an amount according to phase difference of the first and second output signals - being dependent from the physical distance p_P of said two transducers - multiplied by a constant factor not equal to unity or by a factor which is a function of frequency, and generating the result signal from at least one of the first and second output signals and the third signal, which latter represents an acoustical signal as it would be received if a transducer was placed at a distance from one of the provided transducers, which is given by the predetermined factor.

Thereby, there is introduced a virtual distance p_V of transducers which becomes clearly preferable, as for hearing aid applications,

$$p_V > p_P. \tag{9}$$

Thereby, according to formula (8), f_r is shifted to lower frequencies.

Thereby, it becomes possible to realise f_r values well in the audio-frequency band for speech recognition (< 4 kHz) with physical distances of microphones, which are considerably smaller than this was possible up to now.

Multiplying the phase difference by a constant factor does nevertheless not affect the roll-off according to fig. 4. This roll-off is significantly improved, leading to an enlarged frequency band B_r according to fig. 4 if the predetermined function of frequency is selected as a function which is at least in a first approximation inversely proportional to the frequency of the acoustic signal.

For instance for the first order cardoid according to fig. 3 and fig. 4, there may be reached a flat frequency characteristic between 0,5 and 4 kHz and thus a significantly enlarged frequency band B_r with well-defined roll-offs of amplification at lower and higher frequencies by accordingly selecting the frequency dependent function to be multiplied with the phase difference.

To fulfil the above mentioned object there is further provided a hearing aid apparatus which comprises at least two acoustical/electrical transducers spaced from each other by a predetermined distance, whereby the at least two transducers generate, respectively, first and second electrical output signals and wherein the outputs of the acoustical/electrical transducers are operationally connected to a signal processing unit which generates a result output signal from the first and second output signals of the transducers, which result output signal being a function of acoustical signals received at least substantially simultaneously at the two transducers and amplified by a spatial reception characteristic of amplification as a function of spatial angle under which the acoustical signals impinge on the two transducers and which further comprises

- a phase difference detection unit operationally connected to the outputs of the two transducers and generating a phase difference indicative signal; and wherein
- the output of the phase difference detecting unit is operationally connected to the first input of a multiplier unit, the second input thereof being connected to the output of a function or constant generator unit, the output of the multiplication unit being operationally connected to the control input of a phase shifter unit with a signal input operationally connected to the output of one of the transducers, the output of the phase shifter unit and at least one of the outputs of the at least two transducers being operationally connected to the processing unit.

Other objects of this invention will become apparent as the description proceeds in connection with the accompanying drawings, of which show:

Fig. 1: A functional block diagram of a two-transducer acoustic receiver with directional beam forming according to prior art;

Fig. 2: one of prior art beam forming techniques as may be incorporated in the apparatus of fig. 1, shown in block diagram form;

Fig. 3: a two-dimensional representation of a three-dimensional cardoid beam, i.e. amplification characteristic as a function of incident angle of acoustical signals;

Fig. 4: the frequency dependency of the maximum amplification value according to fig. 3 for first and second order cardoid functions;

Fig. 5: a pointer diagram resulting from the technique according to fig. 2, still prior art;

Fig. 6: a pointer diagram based on fig. 5 (prior art), but according to the inventive method, which is performed by an inventive apparatus;

Fig. 7: a simplified block diagram of an inventive apparatus, especially of an inventive hearing aid apparatus, wherein the inventive method is implemented.

As was mentioned above, in the figs. 1 to 4 known beam forming techniques based on at least two acoustical/electrical transducers spaced from each other by a predetermined distance p_P have been explained.

In fig. 5 there is shown a pointer diagram according to (6).

The basic idea of the present invention shall be explained now with the help of the stil simplified one - ω - frequency example. The inventively realised pointer diagram is shown in fig. 6. The phase difference $\omega \cdot dt$ between signal A_2 and A_1 according to fig. 6 is

$$\omega \cdot dt = \omega \cdot \frac{p_p \sin\theta}{c} = \Delta\varphi. \qquad (10)$$

This phase difference is determined and is multiplied by a value dependent from frequency, thus with the respective value of a function $M(\omega)$, which may be also a constant M_o # 1.

By phase shifting one of the two signals A_1, A_2 according to the respective pointers in fig. 6, e.g. of A_2 by

$$M(\omega) \cdot \Delta\varphi \text{ or by } M_o \cdot \Delta\varphi,$$

there results the phase shifted pointer A_{2V}. This pointer would have also occurred if dt had been larger by an amount according to $M(\omega)$ or M_0, thus if a "virtual transducer" had been placed distant from transducer 1_a by the virtual distance

p_V, for which:

$$p_V = M(\omega) \cdot p_P \text{ or} \tag{11}$$

$$p_V = M_0 \cdot p_P. \tag{12}$$

As we consider one single frequency for simplicity we may write $M_0 = M_\omega$.
With virtual τ_V

$$\tau_V = M_\omega \cdot \tau \text{ and} \tag{13}$$

$$(3_v)\quad dt_v = M_\omega \cdot p_P \frac{\sin\theta}{c}$$

we get according to the present invention:

$$(1_v)\quad A_1 = A_{1v} = A\sin\omega t$$

$$(2_v)\quad A_{2v} = A\sin\omega(t+dt_v) = A\sin\omega(t+M_\omega dt)$$

$$(5_v)\quad A_{1v} = A\sin\omega(t+M_\omega\tau)$$

$$(6_v)\quad A_{rv} = 2A\sin((M_\omega \cdot \omega(\tau-dt)/2)$$

With (8) we further get:

$$f_{rV} = \frac{c}{4M_\omega P_p} = \frac{1}{M_\omega} \cdot f_r.$$

Therefrom, we may see that for a given p_P which would lead to a too high f_r, f_{rV} is reduced by the factor $\frac{1}{M\omega}$, taken $M_\omega > 1$. In fig. 7 there is schematically shown a preferred realisation form of an inventive apparatus in a simplified manner, especially for implementing the inventive method into an inventive hearing aid apparatus. Thereby, the output signals of the acoustical/electrical transducer 2_a and 2_b are fed to respective analogue to digital converters 20a, 20b, the outputs thereof being input to time domain to frequency domain - TFC - converter units as to Fast-Fourier Transform units 22a and 22b. A spectral phase difference detecting unit 27 spectrally detects phase difference $\Delta\varphi_n$ for all n spectral frequency components which are then multiplied by a set of constants c_n. If $M(\omega)$ is valid, then the c_n can be different for different frequencies, and represent a frequency dependent function or factor. If on the other hand the phase differences $\Delta\varphi_n$ are multiplied by the same $c_0 = c_n \# 1$ this accords with using a constant M_0.

This multiplication according to (3_V) is done at a spectral multiplication unit 28. Signal A_1 in its spectral representation is then spectrally phase shifted at a spectral phase shifter unit 29 by the multiplied spectral phase difference signals output by multiplier unit 28.

According to fig 7 the signal A_1 in its spectral representation and inventively, spectrally phase shifted is computed in a spectral computing unit 23 together with A_2 in its spectral representation, as if transducer 2a was distant from transducer 2b by a distance $p_V = M(\omega)p_P$. The resulting spectrum is transformed back by a frequency to time domain converter - FTC - as by an Inverse-Fast-Fourier-Transform unit 24 to result in $A_{r\#}$.

Thereby, other beam forming techniques than that described with the help of figs 1 to 4, i.e. using the time delaying technique - transformed in the frequency domain - may be used in unit 23.

Nevertheless the time delaying technique is preferred.

With an eye on fig. 4 it has been explained that by inventively introducing a "virtual" transducer with a virtually enlarged distance from one of the two physically provided transducers, it becomes possible to shift the high gain frequency f_r towards lower frequencies, which is highly advantageous especially for hearing aid applications. This is already reached if instead of a frequency dependent function $M(\omega)$, a constant M_0 is multiplied with the phase difference as explained.

In a preferred mode of the invention the frequency dependent function $M(\omega)$ is selected to be, at least in a first approximation,

$$M(\omega) \sim \frac{1}{\omega} \tag{14}$$

Thereby, it is reached that different from fig. 4 there will be no roll-off and the gain in target direction will be constant over the desired frequency range. By appropriately selecting the function $M(\omega)$ it is e.g. possible to reach a flat characteristic within a predetermined frequency range, e.g. between 0.5 and 4 kHz with defined roll-offs at lower and higher frequencies. With appropriately selecting the function $M(\omega)$ practically any kind of beam forming can be made.

For generating higher order cardoid weighing functions it is absolutely possible to additionally use the not phase-shifted output signal A_1 - as shown in fig. 7 by dotted line - as computing input signal to unit 23 too, thus "simulating" three transducers.

It is evident for the skilled artisan that

- more than two real transducers may be used and/or
- more than one $M(\omega)$ function may be used to produce more than one virtual transducer signal from one or from more than one real transducer signals respectively.

With selecting the number of physical and virtual transducers the spatial weighing function may be selectively tailored, thereby e.g. to completely suppress signals from unwanted directions, e.g. only to pass signals from within a narrow range of target direction and suppress all others. On the other hand, it is also possible to just suppress unwanted sources, thus realising an in fact omnidirectional behaviour, thereby just suppressing one or more distinct noise sources at predetermined spatial angles.

The present invention under its principal object makes it possible to realise beam forming with at least two transducers separated by only a predetermined small distance, due to the fact that electronically there is provided a virtual transducer distance from one of the two physically provided transducers.

Thereby, roll-off may be significantly reduced by such virtual transducer, which is established with a distance dependent from frequency. For a hearing aid apparatus the real distance between the at least two transducers, i.e. microphones, is selected to be 20 mm at most.

Claims

1. A method for electronically and virtually changing the effective distance between a first and second acoustical/electrical transducer, generating respectively first and second electrical output signals, representing substantially simultaneously impinging acoustical signals, said first and second electrical output signals being electronically treated so as to generate a result signal, said result signal being a function of acoustical signals at least substantially simultaneously received at said two transducers and amplified by a spatial reception characteristic of amplification as a function of spatial angle under which said acoustical signal impinge on said transducers and further comprising the steps of

    - generating a third signal from at least one of said first and second output signals by phase-shifting said at least one of said first and second output signals by an amount given by the phase difference between said two output signals, said phase difference being dependent from the physical distance of said two transducers, multiplied by a constant factor not equal to unity or by a factor which is a function of frequency,
    - generating the result signal from at least one of said first and second output signals and said third signal, which latter representing an acoustical signal as it would be received if a transducer was placed at a distance from one of the provided transducers, which distance being given by said constant or said predetermined factor.

2. The method of claim 1, wherein said first and second output signals are transformed in their respective frequency spectra and wherein said third signal is formed as a frequency spectrum, wherein each spectral component at one of said output signals and at a frequency considered is shifted in phase according to phase difference of the spectral components of said first and second output signals at said frequency considered, said phase difference being multiplied by equal or frequency-specific coefficients.

3. The method of claim 1, wherein said frequency dependent function is selected as inversely proportional to frequency at least in a first approximation.

4. The method of claim 2, wherein said frequency-specific coefficients are selected inversely proportional to the frequency considered.

5. The method of claim 1, further comprising the step of selecting the mutual physical distance of said first and second transducers to be at most 20 mm and selecting said distance given by said factor to be larger than said physical distance and incorporating said first and second transducers into a hearing aid apparatus.

6. A hearing aid apparatus comprising at least two acoustical/electrical transducers spaced from each other by a predetermined distance, whereby the at least two transducers generate, respectively, first and second electrical output signals and wherein the outputs of said acoustical/electrical transducers are operationally connected to a signal processing unit which generates a result output signal from said first and second output signals of said transducers, which result output signal being a function of acoustical signals received at least substantially simultaneously at said at least two transducers and amplified by a spatial reception characteristic of amplification as a function of spatial angle under which the acoustical signals impinge on said at least two transducers, and which further comprises:

- a phase difference detection unit operationally connected to the outputs of said two transducers and generating a phase difference indicative signal and wherein

- the output of said phase difference detection unit is operationally connected to the first input of a multiplier unit, the second input thereof being operationally connected to the output of a function generator or constant-generator unit, the output of said multiplication unit being operationally connected to the control input of a phase shifter unit having a signal input operationally connected to the output of one of said transducers, the output of the phase shifter unit and at least one of the outputs of said at least two transducers being operationally connected to said signal processing unit.

7. The apparatus of claim 6, wherein the outputs of said transducers are operationally connected to respective analogue to digital converters, the outputs of said converters being fed to respective transform units generating output signals representing the input signals to said transform unit in the frequency domain, and further providing said phase difference detection unit, said multiplier unit, phase shifter unit as operating in the frequency domain as well as said signal processing unit and further retransforming the output signal of said processing unit into time domain by means of a frequency to time domain conversion unit.

8. The apparatus of claim 6, wherein said function generator generates said function at least in a first approximation inversely proportional to frequency of said acoustic signals impinging on said at least two transducers.

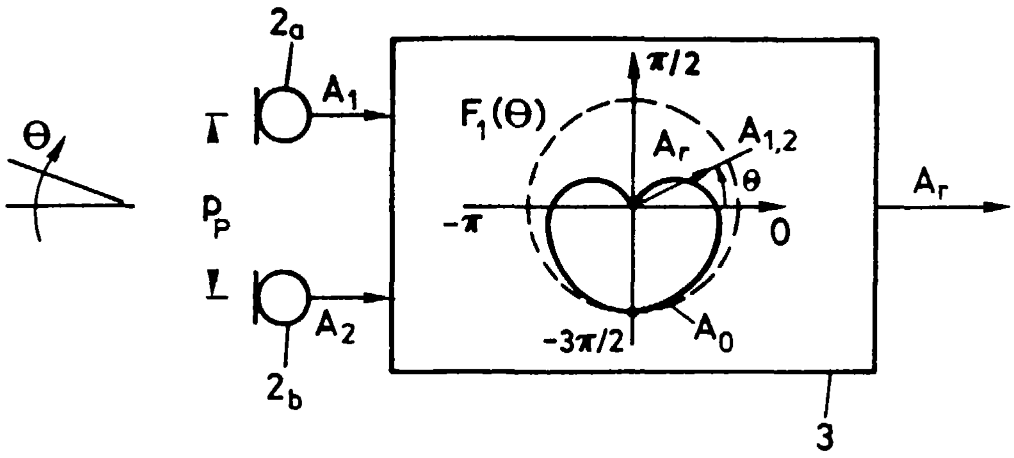

FIG.1

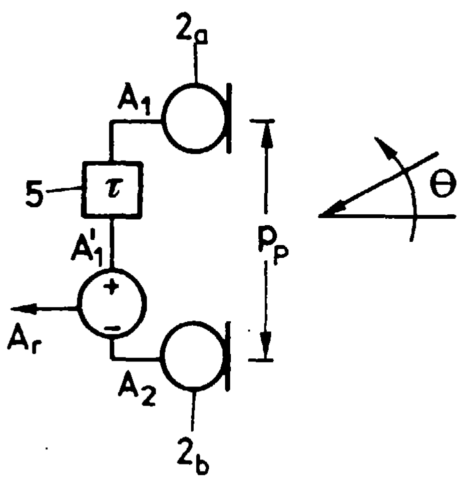

FIG.2

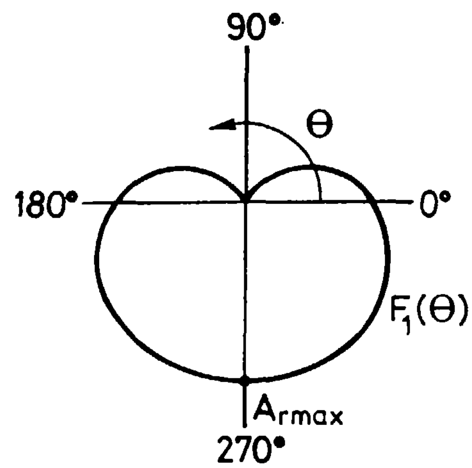

FIG.3

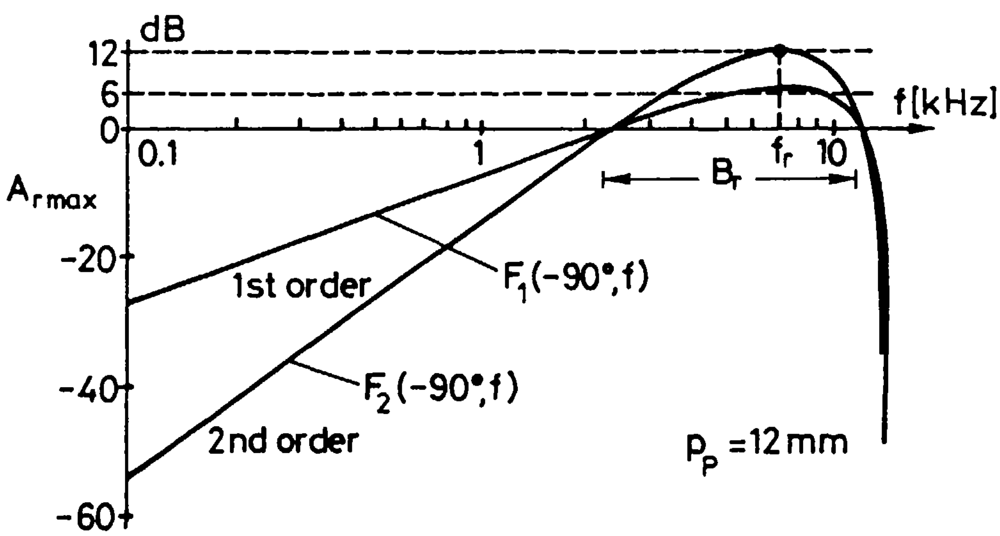

FIG.4

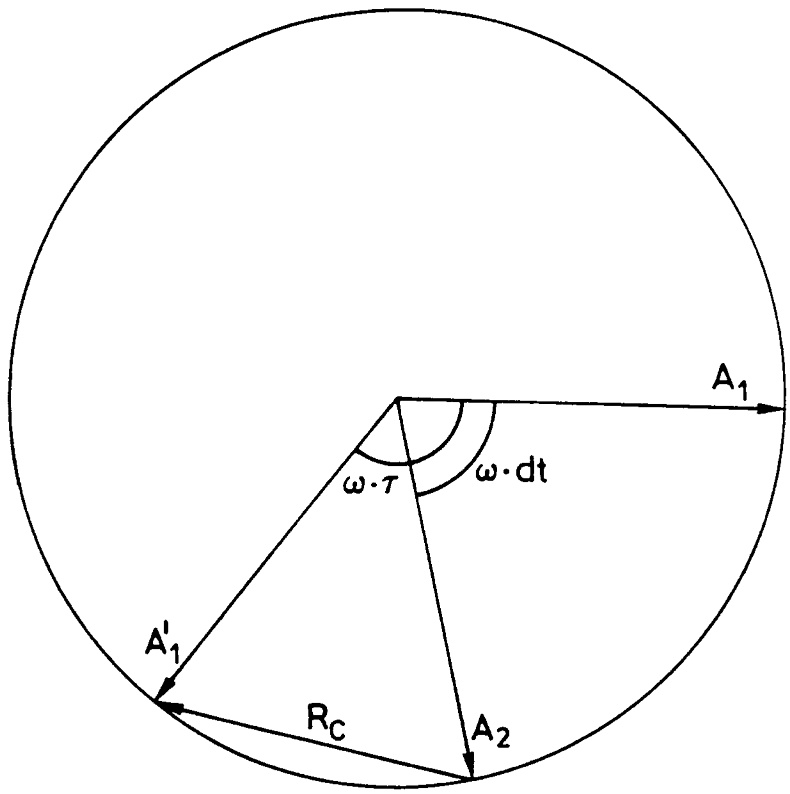

FIG.5

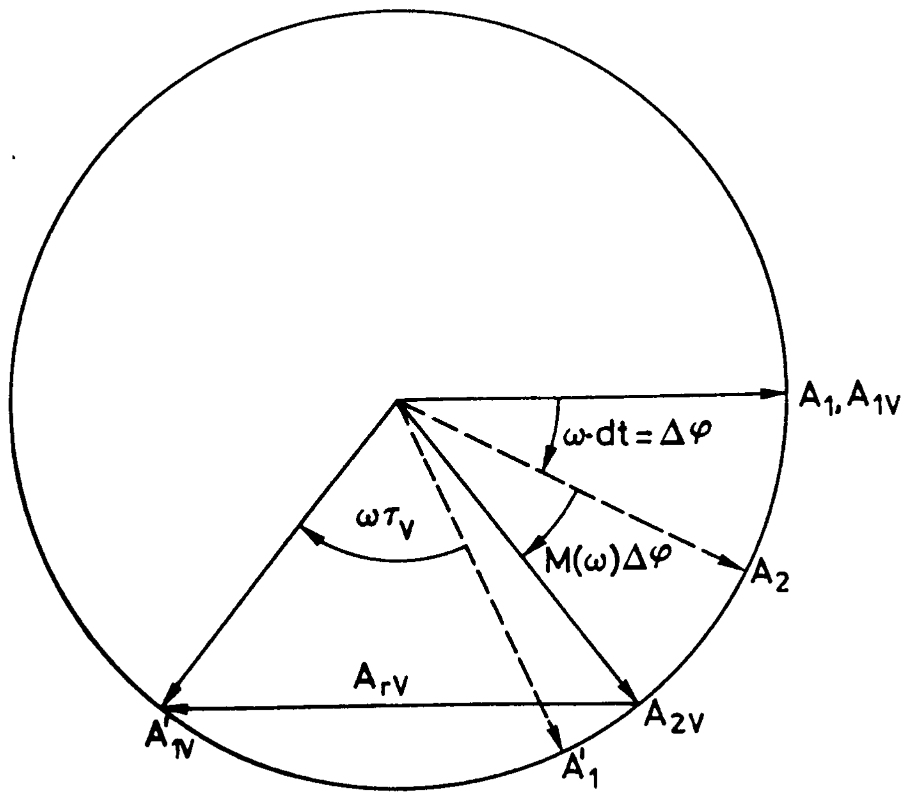

FIG.6

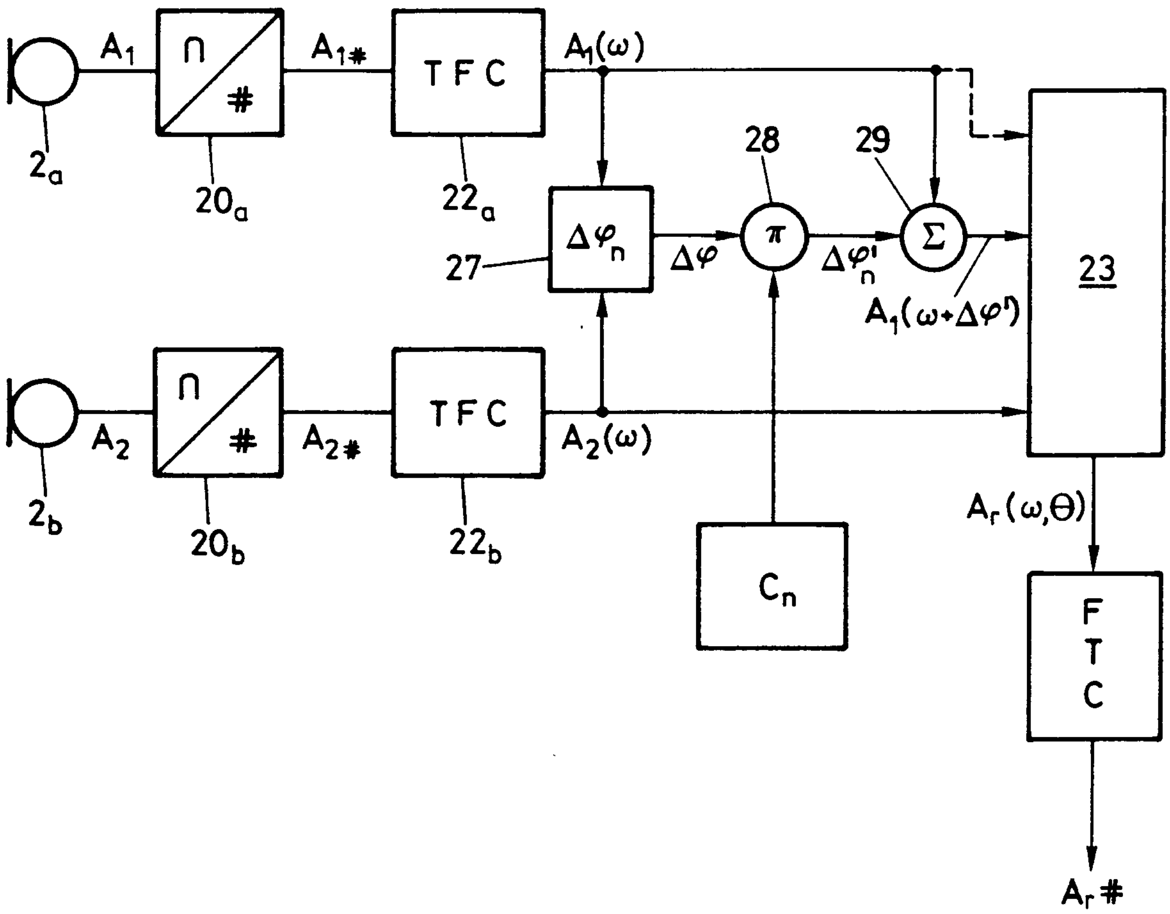
A1
A1#
TFC
A1(ω)
2a
20a
22a
28
29
Δφn
Δφ
π
Δφ'n
Σ
27
A1(ω+Δφ')
23
A2
A2#
TFC
A2(ω)
2b
20b
22b
Cn
Ar(ω,Θ)
F
T
C
Ar#

FIG.7