ENCODING, TRANSMITTING AND RECOVERING SIGNALS

Field of the Invention

This invention relates to the ~~intelligible~~ encoding of signals on carriers, which are transmitted and the ~~encoding~~ signals intelligibly recovered, and more particularly, to the ~~intelligible~~ encoding of speech on a carrier and the intelligible recovery of the speech ~~using~~ (by means of) the Radio Frequency Hearing Effect.

Background

The Radio Frequency ("RF") Hearing Effect was first noticed during World War II as a "click" produced by a pulsed radar signal when the transmitted power is above a "threshold" level. Below the threshold level, the click cannot be heard.

The discovery of the Radio Frequency Hearing Effect suggested that a pulsed RF carrier could be encoded with an amplitude modulated ("AM") envelope. In one approach to pulsed carrier modulation, it was assumed that the "click" of the pulsed carrier was similar to a data sample and could be used to synthesize both simple tones and complex tones such as speech. Although pulsed carrier modulation is suitable for simple tones, it severely distorts the complex tones that form speech, as has been confirmed experimentally.

The presence of this kind of distortion has prevented extending the click process to the encoding of intelligible speech. An example is provided by AM sampled data modulation. Upon demodulation the perceived speech signal has some of the envelope characteristics of an audio signal. Consequently a message can be recognized as speech when a listener is pre-advised that speech has been sent. However, if the listener does not know the content of the message, the audio signal is unintelligible.

The attempt to use the click process to encode speech has been based on the assumption that if simple tones can be encoded, speech can be encoded as well, but this is not so.

Accordingly, it is an object of the invention to provide a novel technique for the intelligible encoding of signals. A related object is to provide for the intelligible encoding of speech.

Another object of the invention is to make use of the Radio Frequency ("RF") hearing effect in the intelligible demodulation of encoded signals, including speech.

A further object is to adapt the "click" effect produced by a radar transmitter signal when the power is above a "threshold" peak level, first noticed during World War II, to the intelligible demodulation of encoded signals, particularly at audio frequencies. such that the modulation will be intelligibly demodulated by means of the RF hearing effect

Still another object of the invention is to suitably encode a pulsed RF carrier with an amplitude modulated ("AM") envelope such that the modulation will be intelligibly de modulated by means of the R.F. Hearing Effect. A related object is to adapt the "click" of a pulsed carrier to function as a data sample in synthesizing simple and complex tones and speech. Another related object is to adapt the click approach to the encoded transmission and recovery of speech without experiencing severe distortion in the recovered speech.

Fig. 2 is spherical demodulator and radiator having a specific acoustic impedance for demodulation using the ~~click~~ RF Hear effect;

Fig. 3 is a diagram illustrating the overall process and constituents of the invention; and

Fig. 4 is an illustrative circuit and wiring diagram for the components of Fig. 3.

Detailed Description

With reference to the drawings, Fig 1 illustrates the RF to acoustic demodulation process of the invention. Ordinarily an acoustic signal A reaches the outer ear E of the head H and traverses first to the inner ear I and the to the acoustic receptors of the brain B. A modulated RF signal, however, enters a demodulator D, which is illustratively provided by the mass M of the brain, and is approximated, as shown in Fig. 2, by a sphere S of radius r in the head H about. The radius r of the sphere S is about 7 cm to make the sphere S equivalent to about the volume of the brain B. It will be appreciated that where the demodulator D, which can be an external component, is not employed with the acoustic receptors of the brain B, it can have other forms.

The sphere S, or its equivalent ellipsoid or similar solid, absorbs RF power which causes an increase in temperature that in turn causes an expansion and contraction which results in an acoustic wave. As a first approximation, it is assumed that the RF power is absorbed uniformly in the brain. Where the demodulator D is external to the brain B, the medium can be selected to assure uniform absorption.

For the modulated RF signal of Fig. 1, the power absorbed in the sphere S is proportional to the power waveform of the modulated RF signal. The absorption rate is characterized quantitatively in terms of the SAR (Specific Absorption Rate) in the units of incident watts per kilogram per watt per square centimeter.

The temperature of the sphere S is taken as following the integrated heat input from the power waveform, i.e. the process is adiabatic without loss or gain of heat, at least for short time intervals on the order of a few minutes.

The radial expansion of the sphere follows temperature and is converted to sound pressure, P(t), determined by the radial velocity (U_r) multiplied by the real part of the specific accustic impedance (Z_s) of the sphere, as indicated in equation (1), below.

$$(1) \quad Z_s = p_o c\,(jkr)/(1+jkr) = p_o c\; jf/f_c\,/(1+jf/f_c)$$

Where:

$p_o c$ = density, 1000 kg/m^3 for water

c = speed of sound, 1560 m/s, in water @ 37 ^{o}C

k = wave number, 2π /wavelength

r = sphere radius, in meters (m)

f = audio frequency

f_c = lower cutoff break frequency, $=c/(2\pi r)$

j = the 90 degree phase-shift operator

The specific acoustic impedance for a sphere of 7 cm radius, on the order of the size of the brain, has a lower cut-off break frequency of 3,547 Hertz (Hz) for the parameters given for equation (1). The frequency range of speech is about 300 to 3000 Hz, i.e., below the cut-off frequency. It is therefore the Real part (Re) of Z_s times the radial particle velocity (U_r) which determines the sound pressure, P(t). The real part of Z_s is given by equation (1a), below:

$$\text{(1a)} \qquad \mathrm{Re}\,(Z_s = p_o c\, c(f/f_c)^2/(1+(f/f_c)^2)$$

In the speech spectrum, which is below the brain cut-off frequency, the sphere S is an acoustic filter which "rolls off", i.e. decreases in amplitude at -40dB per decade with decreasing frequency. In addition to any other demodulation processes to be analyzed below, the filter characteristics of the sphere will modify the acoustic signal with a 40dB per decade slope in favor of the high frequencies.

<u>Results for an AM Modulated Single Tone.</u> ω

An RF carrier with amplitude A_c at frequency W_c is AM modulated 100 percent with a single tone audio signal at frequency

ω_a. The voltage (time) equation of this modulated signal is given by equation (2), below:

$$(2) \quad V(t) = A_c \sin(\omega_c t)(1+\sin(\omega_a t))$$

The power signal is $V(t)^2$ as given by equation (3), below:

$$(3) \quad P(t) = A_c^2 \,[3/4 + \sin(\omega_a t) - 1/4 \cos(2\omega_a t) - 3/4 \cos(2\omega_c t) - \cos(2\omega_c t)\sin(\omega_a t) + 1/4 \cos(2\omega_c t)\cos(2\omega_a t)]$$

To find the energy absorbed in the sphere, the time integral of equation (3) is taken times the absorption coefficient, K. The result is divided by the specific heat, SH, to obtain the temperature of the sphere and then multiplied by the volume expansion coefficient, M_v to obtain the change in volume. The change in volume is related to the change in radius by equation (4), below:

$$(4) \quad dV/V = 3\,dr/r$$

To obtain the amplitude of the radius change, there is multiplication by the radius and division by three. The rms radial surface velocity, U_r is determined by multiplying the time derivative by r and dividing by $2^{1/2}$ $\sqrt{2}$. The result, U_r, is proportional to the power function, P(t) in equation (5), below.

single tone, the pressure wave audio signal will consist of the audio tone and a second harmonic at about -6 dB, if the specific acoustic impedances are the same. However, from equation (1) the break frequency of a model 7cm sphere is 3,547Hz. Most of the speech spectrum is below this frequency therefore the specific acoustic impedance is reactive and the real component is given by equation (8a), below:

$$(8a)\quad R_e(Zs(f)) = \rho_o c\,(f/f_c)^2 \,/\, (1 + (f/f_c)^2)$$

Below the cutoff frequency the real part of the impedance varies as the square of the frequency or gives a boost of 40dB per decade. Therefore, if the input modulating signal is 1kHz, the second harmonic will be have a boost of about 4 times in amplitude, or 12dB, due to the variation of the real part of the specific acoustic impedance with frequency. So the second harmonic pressure term in equation (8) is actually four times the power or 6dB higher than the fundamental term. If the second harmonic falls above the cutoff frequency then the boost begins to fall back to ~~1~~ 0 dB. However, for most of the speech spectrum there would be a severe distortion and strong boost of the high frequency distortion components.

Results for Two Tone AM Modulation Analysis.

Because of the distortion attending single tone modulation, predistortion of the modulation ~~can~~ could be attempted such that the resulting demodulated pressure wave will not contain harmonic distortion. This will not work, however, because the cross-products of two-tone modulation are quite different from single tone as shown below.

Nevertheless, Two-Tone Modulation distortion provides an insight for the design of a corrective process for a complex modulation signal such as speech. The nature of the distortion is defined in terms of relative amplitudes and frequencies.

Equation (8b) is that of an AM modulated carrier for the two-tone case where ω_{a1} and ω_{a2} are of equal amplitude and together modulate the carrier to a maximum peak value of 100 percent. The total modulated RF signal is given by equation (8b), below:

$$\text{(8b)} \quad V(t) = A_c \sin(\omega_c t)[1+1/2 \sin(\omega_{a1} t) + 1/2 \sin(\omega_{a2} t)]$$

The square of (8b) is the power signal, which has the same form as the particle velocity, $U_r(t)$, of equation (9), below. From the square of (8b) the following frequencies and relative amplitudes are obtained of the particle velocity wave, e.g., ω_{a1} which are in the audio range;

$$\text{(9)} \quad U_r(t) = C[\sin(\omega_{a1} t) + \sin(\omega_{a2} t) + 1/4 \cos((\omega_{a1} - \omega_{a2})t) + 1/4 \cos((\omega_{a1} + \omega_{a2})t) - 1/8 \cos(2\omega_{a1} t) - 1/8 \cos(2\omega_{a2} t)]$$

If the frequencies in equation (9) are below the cut-off frequency, the impedance boost correction will result in a pressure wave with relative amplitudes given in equation (9a), below:

$$(9a)\quad p(t) = C'[\ \sin(\omega_{a1} t) + b^2 \sin(\omega_{a2} t) + (1-b^2)/4\cos((\omega_{a1} - \omega_{a2})t) + (1+b^2)/4\cos((\omega_{a1} + \omega_{a2})t) - 1/2\cos(2\omega_{a1} t) - b^2/2\cos(2\omega_{a2} t)$$

where: $b = \omega_{a2}/\omega_{a1}$, and $\omega_{a2} > \omega_{a1}$

Equation (9a) contains a correction factor, b, for the specific acoustic impedance variation with frequency. The first two terms of (9a) are the two tones of the input modulation with the relative amplitudes modified by the impedance correction factor. The other terms are the distortion cross products which are quite different from the single tone distortion case. In addition to the second harmonics, there are sum and difference frequencies. From this two-tone analysis it is obvious that more complex multiple tone modulations, such as speech, will be severely distorted with even more complicated cross-product components. This is not unexpected since the process which creates the distortion is nonlinear.

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This leads to the conclusion that a simple passive predistortion filter will not work on a speech signal modulated on an RF carrier by a conventional AM process, because the distortion is a function of the signal by a nonlinear process.

However, the serious distortion problem can be overcome by means of the invention which exploits the characteristics of a different type of RF modulation process in addition to special signal processing.

AM Modulation With Fully Suppressed Carrier for the Intelligible Encoding of Speech by the Invention for Compatibility With the RF Hearing Phenomena.

The equation for AM modulation with a fully suppressed carrier is given by equation (10), below:

$$(10) \quad V(t) = a(t) \sin(\omega_c t)$$

This modulation is commonly accomplished in hardware by means of a circuit known as a balanced modulator, as disclosed, for example in "Radio Engineering", Frederick E. Terman, p. 481-3, McGraw-Hill, 1947.

The power signal has the same form as the particle velocity signal which is obtained from the square of equation (10) as shown in equation (11), below:

$$(11) \quad P(t) = C\,U_r = a(t))^2/2 - (a(t))^2/2)\cos(2\omega_c t)$$

From inspection of equations (10) and (11) it is seen that, if the input audio signal, a(t), is pre-processed by taking the square root and then modulating the carrier, the audio term in the particle velocity equation (11) (except for amplitude scaling) is an exact, undistorted, replication of the input audio signal. Since the audio signal from a microphone is bipolar, it must be modified by adding a very low frequency (essentially d.c.) bias term, A, such that the resultant sum, a(t)+A>0, is always positive. This is necessary in order to insure a real square root. The use of a custom digital speech processor implements the addition of the term A, i.e. as shown in equation (10*), below:

$$(10^*) \quad V(t) = (a(t)+A)^{1/2} \sin(\omega_c t)$$

The pressure wave is given by equation (11*), below:

$$(11^*) \quad P(t) = C U_r = A/2 + a(t)/2 - (a(t)/2) \cos(2\omega_c t) - (A/2) \cos(2\omega_c t)$$

When the second term of the pressure wave of equation (11*) is processed through the specific acoustic impedance it will result in the replication of the input audio signal but will be modified by the filter characteristics of the Real part of the specific acoustic impedance, $R_e(Z_s)$, as given in equation (8a). The first term of equation (11*) is the d.c. bias, which is added to obtain a real square root; it will not be audible or cause distortion. The third and fourth terms of (11*) are a.c. terms at twice the carrier frequency ~~and therefore have an average value of zero power. These terms are also at twice the carrier frequency~~ ~~it therefore not be audible nor cause distortion~~

and therefore will not distort nor interfere with the audio range signal, a(t).

Since the filter characteristic of equation (7) is a linear process in amplitude, the audio input can be predistorted before the modulation is applied to the carrier and then the pressure or sound wave audio signal, which is the result of the velocity wave times the impedance function, $R_e(Z_s)$, will be the true replication of the original input audio signal.

A diagram illustrating the overall system 30 and process of the invention is shown in Fig. 3. The input signal a(t) is applied to an Audio Predistortion Filter 31 with a filter function As(f) to produce a signal a(t)As(f), which is applied to a Square Root Processor 32, providing an output $(a(t)As(f) + A)^{1/2}$, which goes to a balanced modulator 33. The modulation process known as suppressed carrier produces a double sideband output $(a(t)As(f) + A)^{1/2} \sin(\omega t)$, where ω is the carrier frequency. If one of the sidebands is suppressed (not shown) the result is single sideband (SSB) modulation which also has a suppressed carrier and will function in the same manner discussed above, but for the purpose of implementing the invention. However, the AM double sideband suppressed carrier as described is more easily implemented.

The output of the balanced modulator is applied to a spherical demodulator 34, which recovers the input signal a(t) that is applied to the inner ear 35 and then to the acoustic receptors in the brain 36.

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The various components 31-33 of Fig. 3 are easily implemented as shown, for example by the corresponding components 41-43 in Figure 4, where the Filter 41 can take the form of a low-pass filter, such as a constant-k filter formed by series inductor L and a shunt capacitor C. Other low-pass filter are as shown, for example, in the ITT Federal Handbook, 4th Ed., 1949. As a result the Filer output is $As(f) \propto 1/f^2$. The Root Processor 42 can be implemented by any square-law device, such as the diode D biased by a battery B and in series with a large impedance (resistance) R, so that the voltage developed across the diode D is proportional to the square root of the input voltage a(t)As(f). The balanced modulator 43, as discussed in Terman, ~~supra~~ opere citato (opere cita op. cit / op. cit.), has symmetrical ~~active elements~~ diodes A1 and A2 with the modulating voltage M applied in opposite phase to the ~~active element~~ diodes A1 and A2 through an input transformer T1, with the carrier W applied commonly to the ~~active elements~~ diodes in the same phase, while the modulating signal is applied to the ~~active elements~~ diodes in opposite phase so that the carrier cancels in the primary of the output transformer T2 and secondary output is the desired double side band output.

Finally the Spherical Demodulator 45 is the brain as discussed above, or an equivalent mass that provides uniform expansion and contraction due to thermal effects of carrier energy.

also note diodes are passive elements not active

(* also note diodes are passive elements not active)

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The invention provides a new and useful encoding for speech ~~of~~ on an RF carrier such that the speech will be intelligible to a human subject by means of the RF hearing demodulation phenomena. Features of the invention include the use of AM fully suppressed carrier modulation, the preprocessing of an input speech signal by a compensation filter to de-emphasize the high frequency content by 40dB per decade and the further processing of the audio signal by adding a bias term to permit the taking of the square root of the signal before the AM suppressed carrier modulation process.

The invention may also be implemented using the same audio signal processing and Single Sideband (SSB) modulation in place of AM suppressed carrier modulation. Conventional AM modulation contains both sidebands and the carrier and is not useful for implementation of the invention. Suppressed carrier AM modulation contains both sidebands and no carrier. SSB modulation contains only one sideband and no carrier. The invention may also be implemented by means of digital signal processing to accomplish the processing of the input (rest cannot be deciphered)

The invention may also be implemented using various degrees of speech compression commonly used with all types of AM modulation. Speech compression is implemented by raising the level of the low amplitude portions of the speech waveform and limiting or compressing the high peak amplitudes of the speech waveform. Speech compression increases the average power content of the waveform and thus loudness. Speech compression can introduce some distortion, so that a balance must be made of the increase in distortion with the increase in loudness to obtain an over-all advantage.

The invention may also be implemented by means of digital signal processing to accomplish the processing of the input through the modulation of ...

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What is Claimed:

1. The method of producing undistorted subjective sound, which comprises the steps of:

pre-processor filtering a modulating signal; and

modulating a fully suppressed carrier by the pre-processor filtered modulating signal.

supposed carrier suppressed carrier

2. The method of claim 1 wherein said carrier is amplitude modulated.

3. The method of claim 1 wherein said pre-processor filtering is of an audio speech signal. Signal

4. The method of using the RF hearing phenomena, comprising the steps of:

providing a model of a radio-frequency to acoustic transducer;

analyzing the model to derive a new modulation process which will permit the RF hearing effect to be used for the transmission of intelligible speech.

5. The method of claim 1 wherein the preprocessing is of a speech input signal to de-emphasize the high frequency content of said signal.

(exactly 40dB is not essential) (exactly 40dB is not essential)

6. The method of claim 5 wherein the preprocessing takes place with a signal reduction of 40dB per decade, more or less. (more or less

7. The method of claim 1 wherein further processing of the signal then takes place by adding a bias and then extracting a root of the waveform.

8. The method of claim 7 wherein the further processing is by taking the square root of said waveform.

9. The method of claim 7 wherein the resultant signal is used to modulate an RF carrier in the AM fully suppressed carrier mode.

10. The method of claim 9 wherein the modulated RF signal is demodulated by an RF to an acoustic process that produces an ~~undistorted~~ intelligible acoustic replication of the original input speech.

11. The method of claim 10 wherein the demodulation is by a thermal to acoustic.

12. The method of claim 10 wherein the demodulation is by energy absorption which causes mechanical expansion in a medium and produces an acoustic signal.

13. The method of claim 12 wherein the demodulation is by energy absorption in an animal head to cause said mechanical expansion and said acoustic signal.

14. The method of claim 12 wherein the expansion in said head produces an acoustic signal which is passed by conduction to an inner ear where said signal is further processed in the same manner as an acoustic signal from an outer ear. .

15. A system for producing a modulated carrier from an input modulating signal, which comprises:

a pre-distortion filter for said input signal; and

means for modulating a fully suppressed carrier by the pre-processor filtered modulating signal.

FIG. 3

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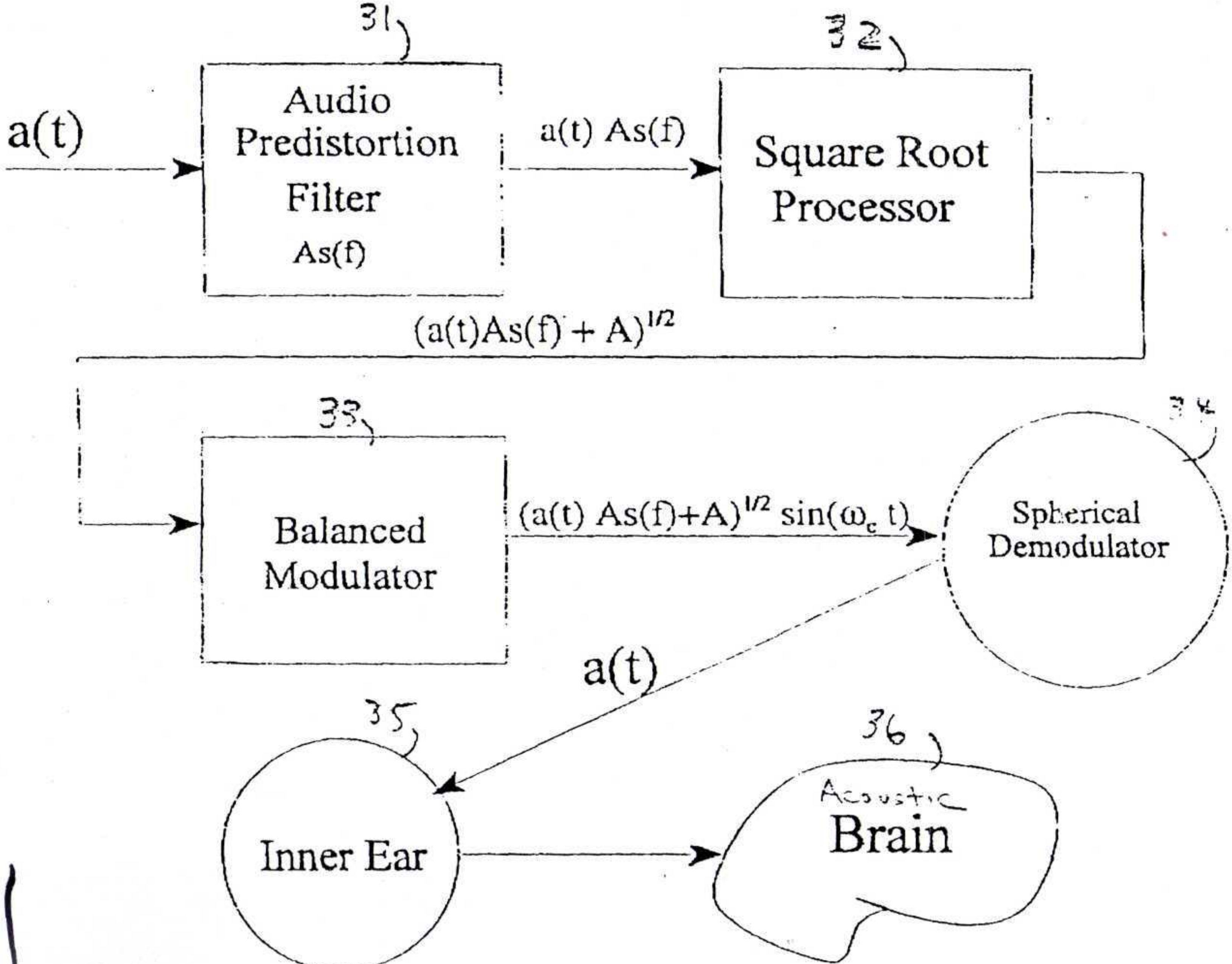

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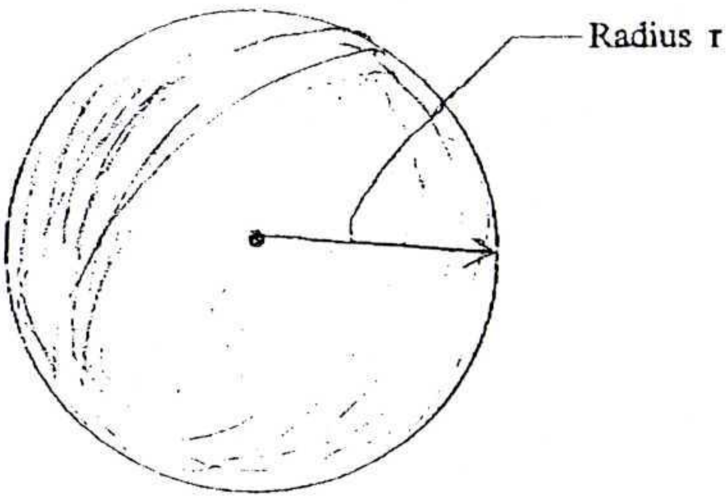

Fig.2 Spherical RF/Acoustic Transducer