From : James O'Loughlin , Diana Loree; Inventors, PL/WSR 01 Nov 94

Subject: Theory and Analysis of RF Hearing, and Invention Disclosure of a Method of Encoding Speech on an RF Signal Which Intelligibly Transmits That Signal to the Hearing Receptors of a Human. (10 pages)

To: File / Record / Distribution

1. **Summary.** Several recent experimental data relative to the RF hearing phenomena indicate that although tones can be encoded there is a serious problem in extending the process to include the encoding of intelligible speech. Clicks and tones can be heard but when speech signals are attempted the intelligibility is practically zero. The original RF hearing phenomena was the "click" produced by a radar signal. It was assumed that the "click" was similar to a data sample and could be used to synthesize tones and speech. The literature also speaks of a "threshold" peak power below which the click could not be heard. This suggested that the encoding of an RF signal should be done as an AM modulated envelope over a pulsed RF carrier train. This approach definitely works for tones but severely distorts speech. In recent experiments during the week of 24 Oct 94, the speech signal perceived has some of the envelope characteristics of the audio signal such that when the subject is told what the speech message was, then he responds that, yes that could have been it. However, if the subject is not told the message content he cannot relate the content. The basic assumption that if one can encode tones, then one can encode speech is not true as shown in the analysis in paragraphs 4 and 5. It is shown that the use of AM fully suppressed carrier modulation with pre-processor filtering of the audio speech input will produce an undistorted subjective sound. This assumes that the model for the RF to acoustic transducer is as described in paragraph 3. .

Using a simple spherical model of the RF power to acoustic transducer, the distortion process was analyzed and predictions are made about the results to be expected with conventional AM modulation and AM fully suppressed carrier modulation. New experiments are in order to better establish the characteristics of the hearing mechanism in order to verify and/or improve the model. Some experiments for future study, research experiments, analysis, and verification are outlined and proposed..

It is concluded that:

(a) Based on the model assumed, it is not possible to encode intelligible speech with the type of AM modulation used in the experiments during the week of 24 Oct 94.

(b) The RF hearing process is an average power driven phenomena and therefore the use of pulse sampling to increase the peak power while holding the average power down is not necessary nor beneficial.

The resulting RF signal will be demodulated by model RF to acoustic process explained below to produce an undistorted acoustic replication of the original input speech signal.

Model Concept. The mechanism for the RF hearing effect has been previously hypothesized by others as a thermal to acoustic demodulation process. We accept this as a basis and include additional refinements. We assume that the process is based on energy absorption in a media in the head which causes a mechanical expansion and thus an acoustic signal. The acoustic signal then is passed by conduction to the inner ear where it is further processed the same as if it were an acoustic signal from the outer ear. See Fig. 1.

The RF to Acoustic Demodulator has characteristics which converts an RF energy input to an acoustic output.

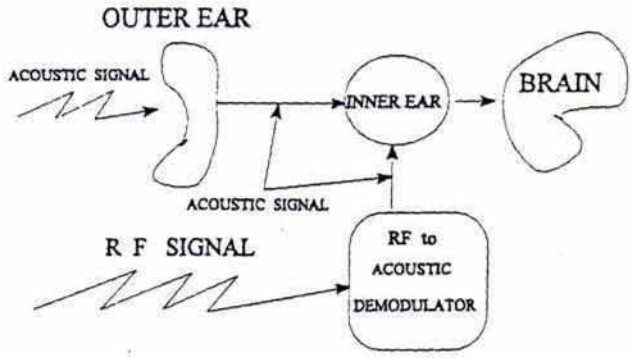


Fig. 1
Model of RF to Acoustic Demodulation Process

The simple model for the demodulator is taken as a sphere of radius r in the head and taken to be about 7cm radius, equivalent to about the volume of the brain. The sphere is assumed to absorb RF power uniformly which causes an increase in temperature which in-turn causes an expansion which results in an acoustic wave. Descriptively, the process begins with an RF signal which is modulated with a signal. The power absorbed in the sphere 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 watts per kilogram per watt per square centimeter incident.

The temperature of the sphere is assumed to follow the integrated heat input from the power waveform, that is, the process is taken to be adiabatic at least for time intervals on the order of a few minutes.

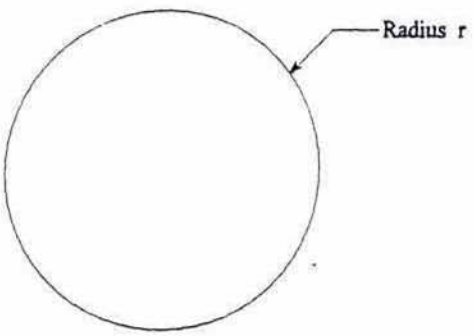


Fig.2 Spherical RF/Acoustic Transducer Model

The radial expansion of the sphere follows the temperature. The radial expansion is converted to a sound pressure, p(t), determined by the radial velocity ($\mathbf{U}_r$) multiplied by the real part of the specific acoustic impedance ($\mathbf{Z}_s$) of the sphere, see Fig.2 and equation (1).

The specific acoustic impedance of the spherical radiator in Fig.2 is ;

(1) $$\mathbf{Z}_s = \rho_0 c\ (j k r)/(1 + j k r) = \rho_0 c\ jf/f_c / (1 + jf/f_c)$$

Where ;

ρ_0 = density, 1000 kg/m^3 for water
c = speed of sound, 1560 m/s, in water @ 37 ^{0}C
k = wave number, $2\pi / \lambda$
r = sphere radius, m
f = audio frequency
f_c = lower cutoff break frequency, = $c/(2\pi r)$

[The sphere size is assumed to be a large volume on the order of the entire brain which corresponds to a sphere of about 7cm radius.]

The specific acoustic impedance for a sphere of 7cm radius has a lower cut-off break frequencies of 3,547 for the parameters given for equation (1). The frequency range of speech is about 300 to 3000 Hz, thus we are working below the cut-off frequency. It is therefore the Real part of $\mathbf{Z}_s$ times the radial particle velocity ($\mathbf{U}_r$) which determines the sound pressure, p(t). The real part of $\mathbf{Z}_s$ is:

(1a) $$\mathcal{R}e(\mathbf{Z}_s) = \rho_0 c(f/f_c)^2 / (1 + (f/f_c)^2)$$

In the speech spectrum, which is below the cut-off frequency, the sphere model is an acoustic filter which rolls of at -40dB per decade with decreasing frequency. So in addition to any other demodulation processes which we will analyze later, the filter characteristics of the sphere will modify the acoustic signal with a 40dB per decade slope in favor of the high frequencies.

Analysis of An AM Modulated Single Tone.

An RF carrier with amplitude A_c at frequency ω_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:

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

The power signal is $V(t)^2$ or :

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

In-order to find the energy absorbed in the sphere, we take the time integral of equation (3) times the absorption coefficient, K. Then divide the result by the specific heat, SH, to obtain the temperature of the sphere and then multiply 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 :

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

To obtain the amplitude of the radius change, we then multiply by the radius and divide by three (3). We now determine the rms radial surface velocity, $\mathbf{U}_r$, by multiplying the time derivitive by r and dividing by $2^{3/2}$. The result, $\mathbf{U}_r$, is proportional to the power function, P(t), in equation (3).

(5) $$\mathbf{U}_r = 0.3535\; P(t)\, r\, K\, M_v \;/\, (3\, SH)$$

To determine the acoustic pressure, p(t), we take the $\mathcal{R}e$ part of the result of multiplying (5) by the specific acoustic impedance (1).

(6) $$p(t) = \mathcal{R}e\,\{\, \mathbf{Z}_s\, \mathbf{U}_r \,\} = \mathcal{R}e\,(\mathbf{Z}_s)\; \mathbf{U}_x$$

The SPL (Sound Pressure Level) , in acoustic dB, is defined as 20 log [p(t)/ 2e-5]. The meaning of the SPL thus determined is somewhat vague. The standard acoustic reference level of 2e-5 Newtons per square meter is based on a signal in air and we are assuming a water like fluid. Therefore, the subjective level in acoustic dB is only approximate, but sufficient for first order accuracy.

In the single tone case the incident RF power, P(t), from equation (3) has two terms which are in the hearing range.

(7) $\sin(\omega_a t) - 1/4 \cos(2\omega_a t)$

To convert this to the acoustic pressure wave,p(t), we must multiply by the specific acoustic impedance calculated at the two frequencies. Therefore, the resulting pressure wave becomes:

(8) $p(t) = C [Zs(\omega_a) \sin(\omega_a t) - 1/4\, Zs(2\omega_a) \cos(2\omega_a t)]$

The result being an audio frequency and a second harmonic at about 1/4 the amplitude. Thus using an RF carrier AM modulated by a 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, this is not the case. From equation (1) the break frequency of the 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 :

(8a) $\mathcal{R}_e (Zs(f)) = \rho_0 c \; (f/f_c)^2 \; / \; (1 + (f/f_c)^2)$

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. However, for most of the speech spectrum we should expect a severe distortion and strong boost of the high frequency distortion components.

The form of this distortion suggests that it might be possible to predistort the modulation in some manner such that the resulting demodulated pressure wave will not contain the harmonic distortion. This scheme will not work because of the cross-products of two tone modulation are quite different from a single tone as shown in the analysis of below.

Two Tone AM Modulation Analysis.

Two-tone modulation distortion is basic to understanding and insight required to design a corrective process for a complex modulation signal such as speech. We now determine what the nature of the distortion is in terms of the 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:

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

The square of (8b) is the power signal, which has the same form as the particle velocity, $U_r(t)$. From the square of (8b) we have the following frequencies and relative amplitudes of the particle velocity wave, which are in the audio range;

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

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

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

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 distorted with even more complicated cross-product components. This is not unexpected since the process which creates the distortion is nonlinear. 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. The serious distortion problem can be over come by means of the invention which exploits the characteristics of a different type of RF modulation process in addition to special signal processing.

Invention of an AM Modulation With Fully Suppressed Carrier Process for the Intelligible Encoding of Speech Which is compatible With the RF Hearing Phenomena.

The equation for AM modulation with a fully suppressed carrier is:

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

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

We have already shown that the power signal has the same form as the particle velocity signal which is obtained from the square of equation (10):

(11) $P(t) = C\ \mathbf{U}_r = a(t))^2/2 - (a(t))^2/2)\cos(2\,\omega_c t)$

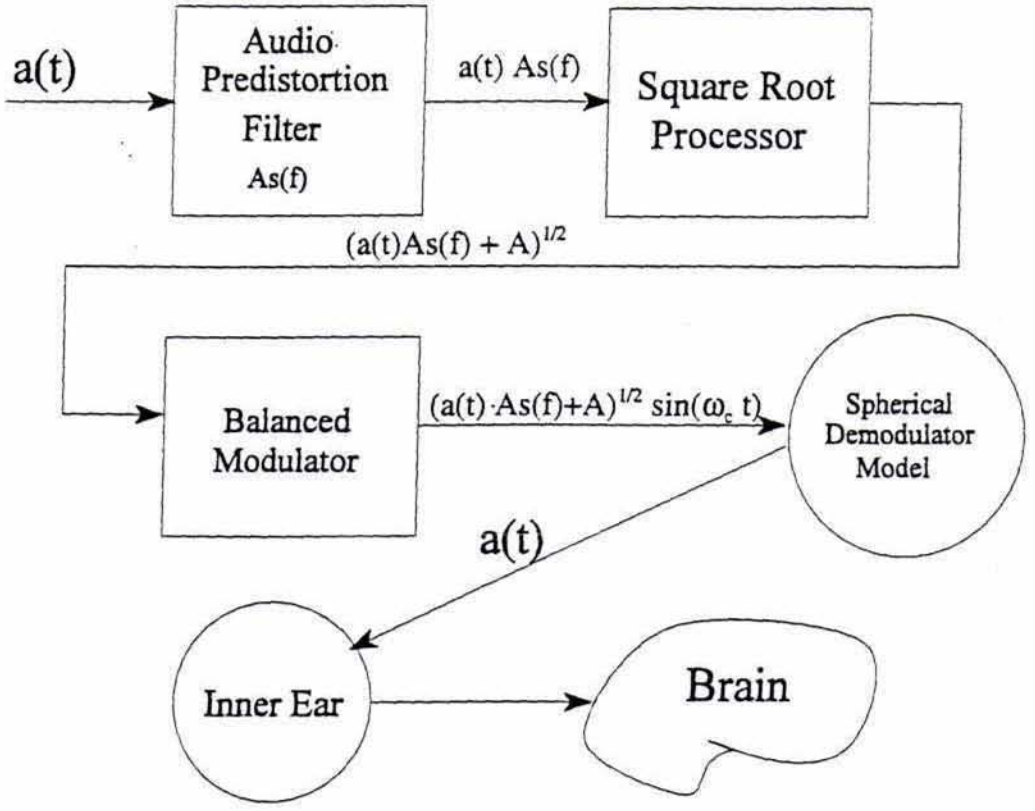
It can be seen by inspection of equations (10) and (11) that, if the input audio signal, a(t), is pre-processed by simply taking the square root of it and then modulating the carrier, the audio term in the particle velocity equation (11) will be an exact, undistorted, replication of the input audio signal. Since the audio signal from a microphone is bipolar, it must be modified by adding a very low frequency (essentially d.c.) 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.:

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

(11*) $P(t) = C\ \mathbf{U}_r = A/2 + a(t)/2 - (a(t)/2)\cos(2\,\omega_c t) - (A/2)\cos(2\,\omega_c t)$

When the second term of the pressure wave of (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, $\mathcal{R}_e(\mathbf{Z}_s)$, as given in equation (8a). The first term of (11*) is the d.c. bias, which was added to obtain a real square root; it will not be audible nor 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 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 we can predistort the audio input, **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, $\mathcal{R}_e(\mathbf{Z}_s)$, will be the true replication of the original input audio signal. The diagram illustrating the invention of the over all process is shown in Fig.3. The modulation process known as single sideband (SSB) will also function in the same manner. However, the AM double sideband suppressed carrier as described is more easily implemented.



Note: $As(f) = 1/f^2$

Fig.3 Block Diagram of the Invention of a Compensated AM Balanced Modulator Process for the Encoding and Transmission of Intelligible Speech Which is Compatible With the RF Hearing Phenomena in Humans

d. ADVANTAGES AND NEW FEATURES OF THE INVENTION

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

e. ALTERNATIVE MODES OF THE INVENTION.

The invention as described 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 using various degrees of speech compression which is commonly used with all types of AM modulation. Speech compression consists of raising the level of the low amplitude portions of the speech waveform and limiting or compressing the high peak amplitudes of the speech waveform. The advantage of speech compression is to increase the average power content of the waveform and thus the loudness. Speech compression does introduce some distortion. In using speech compression one must balance the increase in distortion with the increase in loudness to obtain an over-all advantage.

Attachment to AF Form 1279, Paragraph 12
Description and Operation of the Invention, "A Method for Encoding and Transmitting Speech by Means of the Radio Frequency Hearing Phenomena"

a. PURPOSE of the INVENTION.

The purpose of the invention is to provide a means for the encoding of speech on a radio frequency carrier such that the speech is ineligible to a human subject by means of the RF (Radio Frequency) Hearing Effect.

b. **BACKGROUND.** The RF hearing phenomena was first noticed in the days of World War II, as a "click" produced by a radar transmitter signal. The literature also speaks of a "threshold" peak power below which the click could not be heard. This suggested that the encoding of an RF signal could be done as an AM modulated envelope over a pulsed RF carrier train. One approach to encoding audio signals assumed that the "click" was similar to a data sample and could be used to synthesize tones and speech. This approach works for tones but severely distorts speech. Recent experimental data relative to the RF hearing phenomena show that although tones can be encoded there is a serious problem in extending the process to include the encoding of intelligible speech. In recent experiments at the Air Force Phillips Laboratory during the week of 24 Oct 94, using the AM sampled data modulation process, the speech signal perceived has some of the envelope characteristics of the audio signal such that when the subject is told what the speech message was, then he responds that, yes that could have been it. However, if the subject is not told the message content he cannot relate the content. The basic assumption that if one can encode tones, then one can encode speech is not true; the reason for this is explained later. It is also shown that by using an AM modulation processes with fully suppressed carrier and pre-processor filtering of the audio speech input, will produce an undistorted subjective sound; which is the invention. The explanation of the RF hearing phenomena is based on a simple model for the RF to acoustic transducer.

c. DESCRIPTION, MANNER AND PROCESS OF MAKING AND USING THE INVENTION.

Based on the study and analysis of the RF hearing model, a new modulation process was invented which will permit the RF hearing phenomena to be used for the transmission of intelligible speech. This new process consists of preprocessing the speech input signal with a -40dB per decade filter to de-emphasize the high frequency content of the signal. Then further processing the signal by taking adding a bias and then taking the square root of the waveform. Then by applying the signal to modulate an RF carrier in the AM fully suppressed carrier mode.

6. Invention of an AM Modulation With Fully Suppressed Carrier Process for the Intelligible Encoding of Speech Which is compatible With the RF Hearing Phenomena.

The equation for AM modulation with a fully suppressed carrier is:

(10) $V(t) = a(t) \sin(\omega_c t)$ $\quad (1+a(t)) \sin \omega_c t$

This modulation is commonly accomplished in hardware by means of a circuit known as a balanced modulator. ["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 the square of (10):

(11) $P(t) = C\ \mathbf{U}_r = a(t))^2/2 - (a(t))^2/2) \cos(2\,\omega_c t)$

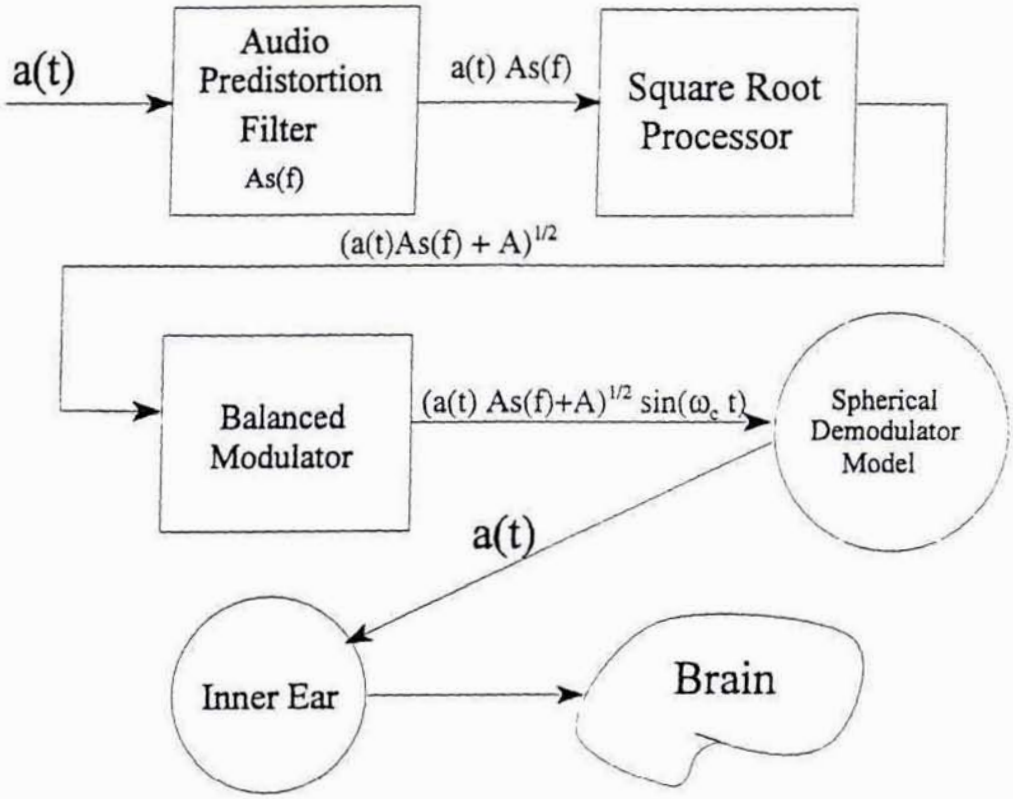
It is obvious from inspection of (10) and (11) that, if we pre-process the signal, a(t), by simply taking the square root of it and then modulating the carrier, the audio term in the particle velocity equation will be an exact , undistorted , replication of the input audio signal. Since the audio signal from a microphone is bipolar, we modify it by adding a very low frequency (essentially d.c.) term such that the result is always positive, $a(t)+A > 0$, 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.:

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

(11*) $P(t) = C\ \mathbf{U}_r = A/2 + a(t)/2 - (a(t)/2) \cos(2\,\omega_c t) - (A/2) \cos(2\,\omega_c t)$

When the second term of the pressure wave of (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, $\mathcal{R}_e(\mathbf{Z}_s)$, as given in equation (8a). The first term of (11*) is the d.c. bias, which was added to obtain a real square root; it will not be audible nor 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 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 we can predistort the audio input, **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, $\mathcal{R}_e(\mathbf{Z}_s)$, will be the true replication of the original input audio signal. The diagram illustrating the invention of the over all process is shown in Fig.3. The modulation process known as single sideband will also function in the same manner. However, the AM double sideband suppressed carrier as described is more easily implemented.



Note: $As(f) = 1/f^2$

Fig.3 Block Diagram of the Invention of a Compensated AM Balanced Modulator Process for the Encoding and Transmission of Intelligible Speech Which is Compatible With the RF Hearing Phenomena in Humans

Observations

Single Tone Experiments. The model used above explains why conventional AM or Pulse Sampled AM tone modulation experiments have been successful. A single tone is replicated as an acoustic pressure wave plus a second harmonic with equal amplitude if both the fundamental and second harmonic are below the cut-off frequency. A subject unless carefully trained and controlled will not notice the second harmonic distortion. However, with controlled reference signals and training it should be possible to identify the existence of the second harmonic in properly designed experiments. Such experiments would provide a good check on the accuracy of the model.

Acoustic Sound Pressure Level (SPL) Prediction. The model formulated can be used to predict the approximate SPL as a function of the radius of the model sphere and the incident modulated RF signal.

As a simple check of the hypothesized process we calculate the sound level based on a single tone audio signal by means of an AM balanced modulator and a 7cm radius spherical demodulator.

We use an incident power density of 100mW/sqcm. The RF carrier is 1GHz, modulated by a 1kHz tone. Using the data from Fig. 8.28 from; USAFSAM-TR-85-73, Radiofrequency Radiation Dosimetry Handbook, Fourth Edition; we estimate a rate of temperature rise of the brain to be about .017 ^{0}C per second. The degrees C per Hz is therefore 7E-4. Since the full peak to peak amplitude occurs in one half cycle we use (17E-6)/2 to calculate the amplitude. The volumetric expansion coefficient is taken to be that of water, 3.5E-4 per unit per degree C. Therefore, the radial displacement is dR = (17E-6)/2 x 3.5E-4 x .07 /3 =1.39E-10m. The rms radial velocity on the sphere is then 1.39E-10 x 1000 / 2.828 = 4.91e-8 m/s. For a 7cm radius sphere the real part of the specific acoustic impedance is $(1000/3547)^2 \rho_0 c / (1 + (1000/3547)^2)$ or 1.23E5 MKS rayls. Thus, the rms pressure is 6.04e-3 Newtons/sqm or a SPL of 49.6 dB. This number is may be artificially high because we use the reference level of 2E-5 Newtons per square meter which is for air whereas the pressure wave was assumed to be in a water like liquid. That is, we have also not accounted for the coupling loss to the inner ear. However, 50 dB is on the order of 10 dB or so higher than the approximate sound level the subjects report. (A normal male voice at one meter is about 50 dB.) The ANSI exposure level at 1GHz is 3mW/sqcm averaged over 6 minutes. Therefore, the 100mW/sqcm level assumed in this calculation could be applied for about 10 seconds within the limits of the standard. The sound level in this example, i.e. using AM balance modulation (suppressed) carrier, will be about 6dB higher than the single tone levels experienced using the pulsed sampling AM modulation in the experiments conducted during the week of 21 Oct 94. The tone amplitude for the AM suppressed carrier case is twice, or 6dB, higher than the conventional or pulse sampled AM modulation signal. A greater advantage is gained when the signal is speech. The suppressed carrier modulation only outputs RF power when the audio signal is present.

The typical audio speech duty factor, about 30 percent, can be used to boost the average power by about 5 additional dB for a possible optimized level of about 50 dB..

If the size of the demodulator is much less than that assumed, i.e. 7cm radius, then the level of the SPL would be greatly reduced. In fact a radius of about 7cm is about optimum for speech. This radius has a cut-off frequency of 3547Hz which is just above the speech spectrum of 300 to 3000Hz. The attenuation of the speech frequencies fall off at 40dB per decade as the cut-off frequency is raised. The cut-off frequency is proportional to the radius, therefore if the radius was 0.7cm the attenuation would be 40dB which would drop the level below that which is sufficient for any practicle use. It is concluded that a volume on the order of the full brain is involved in the demodulation process based on the SPL which is observed. And that the size of the brain of a particular subject will set the lower frequency threshold limit, larger brains will detect lower frequencies than smaller ones.

Page 10 of 10, Subject Title" Theory and Analysis of RF Hearing, and Invention Disclosure of a Method of Encoding Speech on an RF Signal Which Intelligibly Transmits That Signal to the Hearing Receptors of a Human. (10 pages)

INVENTORS; WITNESSETH;

AMES P. O'LOUGHLIN DATE _____ DATE

ANA L. LOREE DATE _____ DATE

TO. George Kersey 06 5 4' 95

ESS / SAZ

FAX 617-377 4820

PAGE 1 of 24

From: Jim O'Loughlin

PL/WSR

KAFB NM

tel 505-846-5497

FAX 505-846-9755