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APPARENT SOUND SOURCE TRANSLATOR

Filed Nov. 19, 1962 3 Sheets-Sheet 1

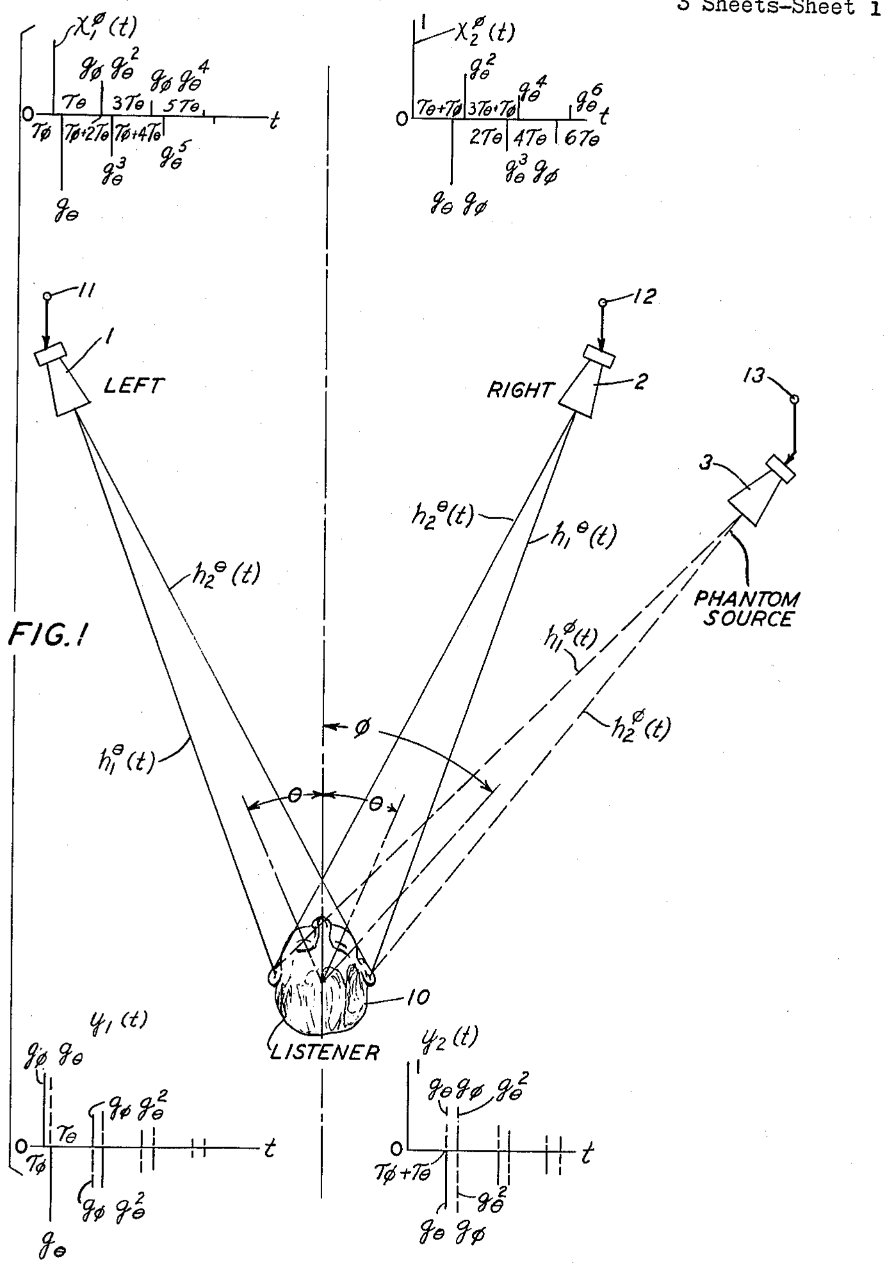

FIG. 1

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FIG. 2

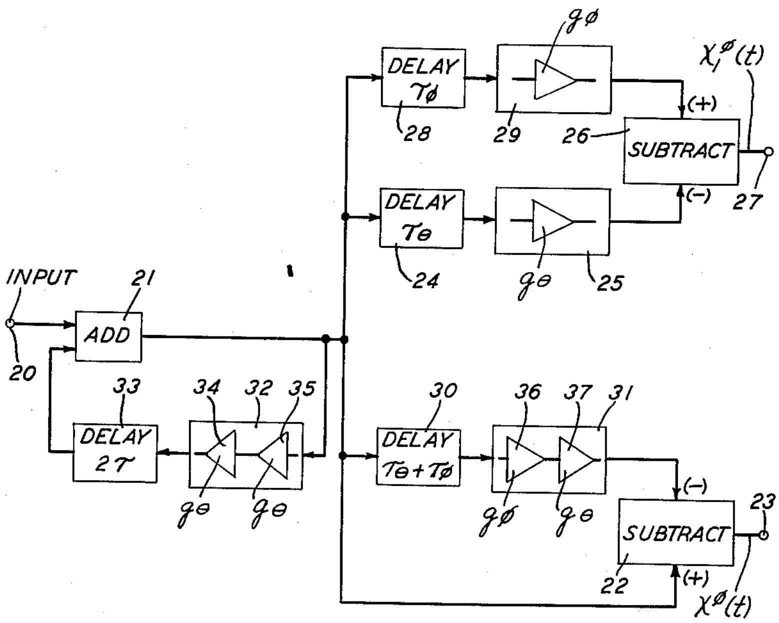


FIG. 4

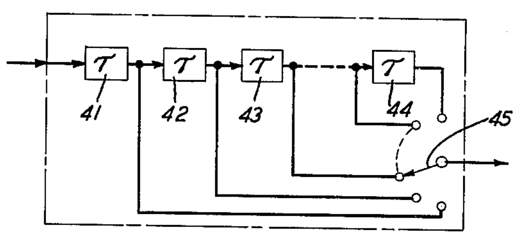


FIG. 3

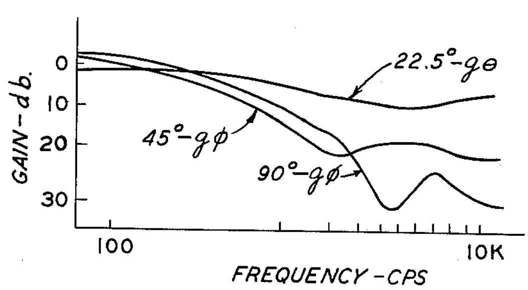


FIG. 5

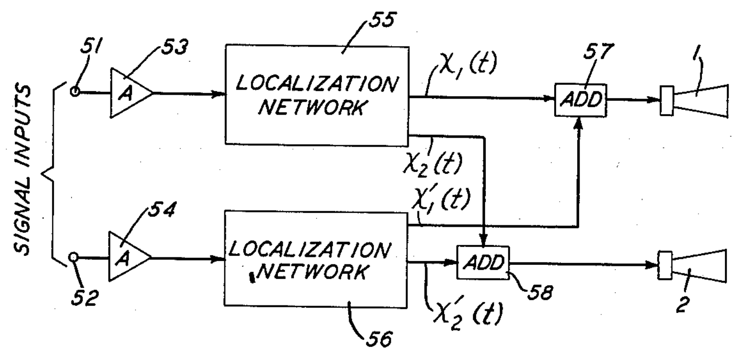
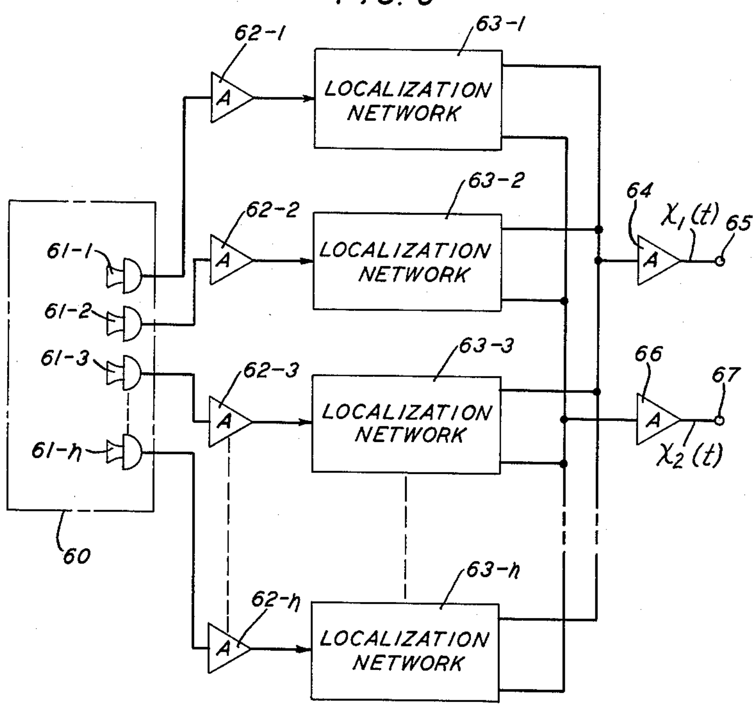

51
53
55
LOCALIZATION NETWORK
$X_1(t)$
57
ADD
1
SIGNAL INPUTS
$X_2'(t)$
$X_1'(t)$
54
LOCALIZATION NETWORK
ADD
58
2
52
56
$X_2'(t)$
A

FIG. 6


62-1
63-1
LOCALIZATION NETWORK
63-2
62-2
LOCALIZATION NETWORK
64
$X_1(t)$
65
61-1
61-2
61-3
61-n
63-3
62-3
LOCALIZATION NETWORK
66
67
$X_2(t)$
60
63-n
62-n
LOCALIZATION NETWORK
A

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3,236,949
APPARENT SOUND SOURCE TRANSLATOR

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9 Claims. (Cl. 179—1)

This invention relates to the reproduction of sound, and in particular to the method of and apparatus for the production of arbitrarily located sound images with only two loudspeakers. It has for its principal object the precise control of the direction of the origin of a so-called "phantom" or virtual sound image in a system in which signals are radiated from two discrete points only. It is another object of the invention to simulate the effect of a number of discrete sound sources placed in different positions by the appropriate energization of two sound sources in fixed positions.

With the advent of high quality stereophonic recording and reproduction, much concern has been given to the exploitation of the spatial illusions which can be created with only two separate loudspeakers supplied with correlated signals. By suitable control of the amplitude and phase of the correlated signals the apparent origin of a sound can be placed arbitrarily at either of the two loudspeakers or at any point between them. Some "widening" of the sound image stage may be achieved by supplying out-of-phase correlated signals to either or both of two radiating loudspeakers. This has the effect of pushing the apparent source further away from the center line of the speaker system. However, in doing this, various undesirable effects are often produced, e.g., various "in the head" localizations are created. While these are not necessarily objectionable, they nevertheless are very often artificial sounding. Another way of broadening the stage involves the use of a great number of loudspeakers individually supplied with signals developed by a corresponding number of spaced microphones. Techniques of this sort yield excellent results, of course, but are economically unattractive because of the great number of individual sound channels and transducer elements required.

This invention attacks the problem of sound localization in a different way. A mathematical analysis of the sound reaching each ear from any arbitrarily placed source is utilized to develop an appropriate "filter" characteristic for each of two channels. Networks with the specified filter characteristic, placed in series with each of two separate loudspeakers, transform the signal radiated by each to one which will produce at both ears of a listener situated at or near the center line between the speakers, a signal identical to one that would be received by him from the arbitrarily placed source. In effect, the signals radiated from the two loudspeakers combine at the listener's ears to produce a pressure wave corresponding to a signal from an arbitrarily located source. To a first approximation, frequency independent network elements create a reasonably sharp virtual sound image. Frequency sensitive networks, however, have been found to produce very sharply defined images in any direction in the sound plane.

The two signals required for the creation of one or more arbitrarily located sound images, in accordance with the invention, contain all of the information on which the listener bases his estimate of the location of the sound. This includes the relative (frequency dependent) amplitude of signals emanating from both loudspeakers, and the times of arrival of the sound pressure waves at each of his ears. The required filter characteristic may

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be imparted to the two correlated signals at any point in a record or reproduce cycle. By use of the apparatus of the present invention at the reproducer station to "filter" signals before they are delivered to two loudspeakers, ordinary sterephonic signals may be given a wider stage width. Alternatively, the filtering may take place at a recording station, to create a pair of recorded signals that contain the required directional information. Independent playback of such a pair of signals in a two-channel system will consequently yield the desired illusion.

The invention will be fully apprehended from the following detailed descriptions of illustrative embodiments thereof taken in connection with the appended drawings in which:

FIG. 1 is a pictorial diagram illustrating the relation to one another of sound pressure waves radiated from two loudspeakers in producing an arbitrarily located sound image in accordance with the invention;

FIG. 2 is a block schematic diagram illustrating localization network apparatus suitable for use in the practice of the invention;

FIG. 3 illustrates suitable frequency characteristics for the frequency dependent amplifiers of FIG. 2 in accordance with the invention;

FIG. 4 is a block schematic diagram showing a multiple-value delay element suitable for use in the apparatus of FIG. 2;

FIG. 5 is a block schematic diagram of apparatus for processing a pair of input signals in accordance with one mode of operation of the apparatus of the invention; and

FIG. 6 is a block schematic diagram of apparatus for processing a plurality of input signals in accordance with another mode of operation of the invention.

Before entering upon a detailed description of the apparatus of the invention and of the fashion in which it operates, it is desirable to discuss certain psychoacoustic (psychological-acoustic) concepts and certain mathematical relations, some of which are implemented by the apparatus shown in the drawings.

The difference in the amplitude and time of arrival of the transients of a sound at a listener's two ears provides most of the information upon which an auditory judgment of the direction of the origin of the sound is based. Since this invention is concerned primarily with localization, the pulse is the best stimulus for analysis. Since the short pulse encompasses all frequencies, it stands as an idealization of transients in all sounds, speech, music, noise, and the like.

The principles of the invention may thus be best described on the basis of single pulses and the manner by which they are used by a listener in determining the origin of a sound. Consider, for example, the arrangement depicted in FIG. 1. A listener **10** faces two loudspeakers **1** and **2** located at equal distances from him and at an angle $\pm\theta$ from the center line of his position. If a signal, e.g., a short pulse $x_1(t)$, is radiated from the left speaker **1**, the sound pressures at the listener's left and right ears will be $h_1^\theta(t)$ and $h_2^\theta(t)$, respectively. Sound pressure wave $h_1^\theta(t)$ will be the stronger of the two and will reach the listener's left ear before sound pressure wave $h_2^\theta(t)$ reaches the right ear. Consequently, the listener will have no difficulty in correctly locating the source of pulse $x_1(t)$ as originating in loudspeaker **1**. Similarly, if a signal, e.g., a short pulse, $x_2(t)$, is radiated from right speaker **2**, the sound pressures at the listener's left and right ears will be $h_2^\theta(t)$, and $h_1^\theta(t)$, respectively. In this case sound pressure wave $h_1^\theta(t)$ will reach the listener's right ear with greater intensity than the sound pressure wave $h_2^\theta(t)$ will reach his left ear. Moreover, it will reach his right ear before it reaches his left ear. Again no difficulty is encountered in correctly locating the origin of the sound as loudspeaker **2**. Clearly then, the listener would imagine a pulse to originate at phantom loudspeaker **3** positioned at an angle φ from the center line if the sound pressure wave perceived at his right ear, e.g., $h_2^{\varphi}(t)$, reached his right ear at a time, depending on the angle φ, before a wave $h_1^{\varphi}(t)$ of a somewhat lesser amplitude reached his left ear. This is precisely what his ear would hear if a sound actually originated at point **3**.

It is in accordance with the present invention to provide at the listener's left and right ears, the appropriate sound pressure waves which would reach his ears from such a source of sound, **3**, from the two fixed position loudspeakers **1** and **2**.

To create the illusion discussed above, e.g., a pulse originating from loudspeaker **3**, a single pulse is initially radiated from loudspeaker **2**. It reaches the listener's right ear at a time which may arbitrarily be designated "zero" time and with a magnitude $g=1$. At a time τ_θ later, the pulse reaches his left ear with a somewhat smaller magnitude g_θ. The delay τ_θ is a function of the angular position of loudspeaker **2**, namely the angle θ. So far as the right ear is concerned, the pulse could well have originated in loudspeaker **3**. However, the attenuated and delayed pulse which reaches the left ear destroys this illusion and positively establishes the source as speaker **2**. Consequently, the pulse reaching the listener's left ear is canceled by a pulse radiated from speaker **1**. Such a pulse must obviously reach the listener's left ear with a magnitude of inverse polarity to the magnitude of the pulse reaching his left ear from speaker **2**, i.e., g_θ, and at a time τ_θ seconds later than the initial pulse from speaker **2** reached his right ear. The illusion is not yet complete since the listener expects in his left ear, a delayed and attenuated pulse from phantom loudspeaker **3**. This pulse also is supplied by loudspeaker **1** at a time τ_ϕ seconds following the reference pulse. With this pulse, the necessary conditions for the establishment of an apparent source at loudspeaker **3** is complete. However, both the pulse from speaker **1** used to cancel the premature pulse from speaker **2**, and the substitute pulse from speaker **1**, reach the listener's right ear. Consequently, speaker **2** must provide the proper signals for canceling these two pulses at the right ear. It will be realized that the magnitudes of these pulses are diminished and that they occur τ_θ seconds following the time of their radiation from loudspeaker **1**. In like manner, the pulses from loudspeaker **2**, used to cancel the unwanted pulses at the listener's right ear, are received τ_θ seconds later in diminished form at his left ear. Once again, loudspeaker **1** provides the necessary canceling signals. Each repetition has an amplitude g_θ^2 times that of the previous pulse and is delayed by a time $2\tau_\theta$.

If the above process is continued for a sufficiently long time, all of the unwanted pulses are eventually canceled out by virtue of the attenuation involved in each subsequent cancellation. All that remains, therefore, at the listener's left and right ears are pulses spaced apart in time and related in amplitude as though they originated in phantom source **3**.

It is thus in accordance with the present invention to energize loudspeakers **1** and **2** with signals which, when combined at the listener's ears, produce pressure waves corresponding to those from any desired direction. Simply expressed, the signals supplied to loudspeakers **1** and **2** are adjusted by means of a suitable localization network in both intensity and phase in conformity with a prescribed schedule.

From the above discussion it may be observed that if $x_1(t)$ and $x_2(t)$ are the inputs required to produce desired pressure responses $y_1(t)$ and $y_2(t)$ at the left and right ears, respectively,

$$y_1(t)=x_1(t)*h_1^{\theta}(t)+x_2(t)*h_2^{\theta}(t) \quad (1)$$

$$y_2(t)=x_1(t)*h_1^{\theta}(t)+x_2(t)*h_2^{\theta}(t) \quad (2)$$

where * signifies convolution. If the Fourier transforms are taken for both sides of Equations 1 and 2 the following relations are obtained:

$$Y_1(\omega)=X_1(\omega)\cdot H_1(\omega)+X_2(\omega)\cdot H_2(\omega) \quad (3)$$

$$Y_2(\omega)=X_1(\omega)\cdot H_1(\omega)+X_2(\omega)\cdot H_2(\omega) \quad (4)$$

where $X_1(\omega)$, $X_2(\omega)$, $Y_1(\omega)$, $Y_2(\omega)$, $H_1(\omega)$ and $H_2(\omega)$ are the Fourier transforms of $x_1(t)$, $x_2(t)$, $y_1(t)$, $y_2(t)$, $h_1^{\theta}(t)$ and $h_2^{\theta}(t)$, respectively. If Equations 3 and 4 are then solved for $X_1(\omega)$ and $X_2(\omega)$, $x_1(t)$, and $x_2(t)$ are obtained by taking the inverse Fourier transforms:

$$x_1(t)=[y_1(t)*h_1^{\theta}(t)-y_2(t)*h_2^{\theta}(t)]*a(t) \quad (5)$$

$$x_2(t)=[y_2(t)*h_1^{\theta}(t)-y_1(t)*h_2^{\theta}(t)]*a(t) \quad (6)$$

where $a(t)$ is the inverse Fourier transform of

$$\frac{1}{H_1^2(\omega)-H_2^2(\omega)}$$

Providing that $a(t)$ is a realizable impulse rseponse, $x_1(t)$ and $x_2(t)$, the required loudspeaker input signals, may be created by implementing Equations 5 and 6. For example, if it is desired to simulate a phantom sound source S at a direction φ from the center line of the listener, it is necessary only that the pressure responses $y_1(t)$ and $y_2(t)$ obtained by radiating a short pulse from the source S be appropriately specified. For simplicity, it may be assumed that the interaural delay and loss due to the sound source in any direction depend only on the angle and not on the frequency of the radiated sound. The pressure responses at the left and right ears of the listener may then be specified as:

$$h_1^{\theta}(t)=g_0\delta(t-\tau_0) \quad (7)$$

$$h_2^{\theta}(t)=g_\theta g_0\delta(t-\tau_0-\tau_\theta) \quad (8)$$

$$y_1(t)=g_\phi g_0\delta(t-\tau_0-\tau_\phi) \quad (9)$$

$$y_2(t)=g_0\delta(t-\tau_0) \quad (10)$$

where g_θ, τ_θ, and g_ϕ and τ_ϕ are the differential gains and delays produced at the two ears by sound incident from directions θ and φ, respectively.

Substituting for $h_1^{\theta}(t)$, $h_2^{\theta}(t)$, $y_1(t)$ and $y_2(t)$ in Equations 5 and 6 the required loudspeaker signals are obtained as follows:

$$x_1(t)=[g_\varphi(t-\tau_\varphi)-g_\theta\delta(t-\tau_\theta)]*\sum_{n=0}^{\infty}g_\theta^{2n}\delta(t-2n\tau_\theta) \quad (11)$$

$$x_2(t)=[1-g_\theta g_\varphi\delta(t-\tau_\theta-\tau_\varphi)]*\sum_{n=0}^{\infty}g_\theta^{2n}\delta(t-2n\tau_\theta) \quad (12)$$

Thus the signals $x_1(t)$ and $x_2(t)$ for the left and right loudspeakers **1** and **2**, respectively, are defined such that the sound pressures at the left and right ears of the listener will be identical to those which would be developed at his left and right ears from a single source radiated from loudspeaker S. Merely by altering the parameters of the signal, the apparent direction of the sound source S may be varied within a half plane, that is, anywhere within approximately 90° of either side of the center line between speakers **1** and **2**.

The required signals $x_1(t)$ and $x_2(t)$ may be produced with the apparatus depicted in FIG. 2. Apparatus of this sort is termed a "localization network." Considering the analysis with a single energizing pulse once again, a single pulse applied to input terminal **20** is passed by way of adder **21** directly to the positive input of subtractor **22** to form the leading pulse of output signal $x_2^{\phi}(t)$. It thus represents the reference pulse which, in the example given above in connection with FIG. 1, denotes the first pulse received from phantom source **3** at position φ degrees to the right of the center line. This pulse reaches the right ear at a time τ_θ later with an amplitude g_θ. Accordingly, the pulse from adder **21** is retarded τ_θ seconds in delay device **24** and conformed in amplitude by amplifier **25** with a gain g_θ to that appropriate to the listener's left ear and supplied to the negative input of subtractor **26**. Data concerning the relative magnitude of g may be obtained from the curves on page 224 of Speech and Hearing in Communication, H. Fletcher, D. Van Nostrand Company, Incorporated, New York, New York, 1953. The value of τ_θ may be derived from experimental data for various values of θ. This delayed and attenuated pulse is available at terminal **27** and, as delivered to loudspeaker **1**, provides the necessary pulse at the listener's left ear for canceling the initial pulse radiated from loudspeaker **2**, e.g., the initial pulse of $x_2^\phi(t)$ available at output terminal **23**. The correct pulse at the listener's left ear is developed by passing the pulse from adder **21** through delay device **28** proportioned to impart a delay t_ϕ to the signal and through amplifier **29** proportioned to attenuate the signal g_ϕ and delivering it to the positive input of subtractor **26**. This component of $x_1^\phi(t)$ is radiated by loudspeaker **1** and reaches the listener's ear at the correct time and with the correct magnitude to supply the necessary directional information by which the listener perceives a signal to originate at phantom source **3**. Since this pulse also reaches the listener's right ear, it is canceled by passing the initial pulse from adder **21** through delay device **30** proportioned with a delay $\tau_\theta+\tau_\phi$ and through amplifier **31** with an attenuation proportional to $g_\theta g_\phi$. A tandem connection of amplifiers **36** and **37**, proportioned respectively with gains g_ϕ (similar to amplifier **29**) and g_θ (similar to amplifier **25**) yields the necessary product signal $g_\theta g_\phi$. This signal is passed with negative polarity from subtractor **22** to terminal **23** and thence to loudspeaker **2**. Since this signal also reaches the listener's left ear, it must be canceled in the left channel. Similarly, all subsequent pulses must be reciprocally canceled indefinitely or until the magnitudes are below the threshold of perception. Since each pulse is attenuated by a factor $-g_\theta$ and delayed by τ_θ when applied to the opposite loudspeaker for canceling purposes, the canceling impulse must be canceled at the first loudspeaker by a pulse of amplitude $(-g_\theta)(-g_\theta)=g_\theta^2$ after a delay of $\tau_\theta+\tau_\theta=2\tau_\theta$. Thus, each pulse must be followed by a pulse attenuated by g_θ^2 and delayed by $2\tau_\theta$. This is achieved by the feedback loop portion of the network of FIG. 2. The initial pulse available at adder **21** is, in accordance with the invention, cycled through amplifier **32** with an attenuation characteristic proportional to g_θ^2 and delay device **33** proportioned to have a delay time $2\tau_\theta$. It is then supplied to adder **21**. Accordingly, every $2\tau_\theta$ seconds following the delivery of the initial impulse to adder **21**, a new pulse is delivered to the circuits previously described. The new pulse, attenuated and delayed as compared with the reference pulse, is further attenuated and delayed in the several branch circuits as described above, so that the ultimate signals delivered to terminals **23** and **27** have the characteristics described above in Equations 11 and 12. Amplifier **32** typically includes the tandem connections of amplifiers **34** and **35**, each with gain g_θ (to yield the requisite gain g_θ^2). Other arrangements for developing a signal proportioned to g_θ^2 may, of course, be used.

It will be realized that, in the previous discussion, the effects of frequency have been ignored. It has been found, however, that even without frequency dependent compensation, remarkably good localization may be obtained with the apparatus of FIG. 2 with the several parameters proportioned as described. In particular, for angles less than about 60° the phantom source is quite apparent for most listeners even with frequency independent amplifiers. For angles greater than about 60°, e.g., $\varphi>60°$, localization is less well defined since it is more strongly dependent on frequency. It is in accordance with the present invention, therefore, to adjust the frequency response of the amplifiers **25**, **29**, **31**, and **32** in the apparatus of FIG. 2 in accordance with the particular angle φ involved, thereby to compensate for the fall-off in sharpness of the phantom source at certain (higher) frequencies.

The frequency response of the several amplifiers may be adjusted in any conventional manner; for example, suitable filters in series with the amplifiers may be used. Alternatively, the response of the amplifiers themselves may be suitably tailored using any of the techniques well known to those skilled in the art. FIG. 3 illustrates several frequency response curves for the amplifiers for various angles. For an angle $\varphi=22.5°$ for example, the required response is relatively flat for all amplifiers. For an angle $\varphi=45°$, it has been found experimentally that the response of amplifiers used to adjust signals according to g_θ may be relatively flat, similar to the curve g_θ shown for 22.5°. The characteristic of amplifiers used to adjust signals to g_ϕ, however, falls off rapidly in frequency, as shown. Similarly, for $\varphi=90°$, tests have shown that the amplifiers of gain g_θ may retain a relatively flat response, and the response of those of gain g_ϕ should, for best directional effects, be suitably attenuated at the higher frequencies.

For improved localization at larger angles, the delays involving τ_ϕ must also be properly adjusted. The correct delay value may be selected, for example, from among several available delay values by means of apparatus connected in the fashion illustrated in FIG. 4. In the figure, fixed delay elements **41**, **42**, **43**, . . . **44** are tandemly connected to provide n different delay values (multiples of τ) at the terminals of a multiple contact switch **45**. τ may typically be 100 microseconds.

The localization network of FIG. 2 may thus be used as a filter to prepare suitable signals from a single one supplied to terminal **20**, for energizing a pair of spaced loudspeakers in a fashion to create the necessary psychological information by which a listener imagines a sound to originate in a phantom source at any desired location in a 180° plane. Although described on the basis of a single pulse, the desired spatial impression created as the amplifier and delay elements of the localization network are properly adjusted, will, as mentioned above, be equally well defined in the case of any real sound source, e.g., speech or music. In general, localization will be good for signals with transients. But, even for the extreme case of a source emitting a sine wave (in which case localization may be difficult for both real and phantom sources depending on the frequency of the sine wave and the reverberation of the listening area), the phantom source created by signals developed in the localization network will be as good (sharp) as for the real source. This result is, of course, to be expected because, if a pulse is reproduced correctly at the ears, it follows from the uniqueness of Fourier transformation that all of its individual frequency components will also be correctly reproduced.

If the network is placed between a microphone or recording transducer and a pair of loudspeakers, the recorded material may be made to appear to originate at a desired phantom location. If the recorded material represents two signals of a stereophonic recording, the apparent separation afforded by the stereophonic recording may be considerably enhanced by use of the localization networks in accordance with the present invention. The latter arrangement is illustrated in FIG. 5. In the figure, individual channel signals of a two-channel stereophonic recording are supplied respectively to input terminals **51** and **52** and, if desired, are adjusted in gain in ordinary linear amplifiers **53** and **54**. The signal developed at the output of amplifier **53** is passed through localization network **55** to produce two output signals $x_1(t)$ and $x_2(t)$. The channel signal at the output of amplifier **54** is passed through localization network **56** to develop a pair of localized signals $x_1'(t)$ and $x_2'(t)$. Localization networks **55** and **56** are of the construction shown in FIG. 2. As described above, the paired output signals contain the necessary clues to psychologically direct a listener's attention to a particular phantom location. The x_1 component signals from networks **55** and **56** are combined in adder **57** and delivered to loudspeaker **1**. Similarly, the $x_2(t)$ signals from the networks are combined in adder **58** and delivered to loudspeaker **2**. By suitably selecting the gain and delay parameters of the networks **55** and **56**, any desired degree of spatial spreading of the sounds emanating from loudspeakers **1** and **2** may be secured. If network **55**, for example, is adjusted to have a plus 90° characteristic and network **56** adjusted to have a minus 90° characteristic, the stereo signals, which normally would be made to appear to originate from the two loudspeaker positions or from position between them, may be made to appear to originate from points in a 180° area, i.e., from anywhere in a half plane.

On occasion it is also sometimes desirable to create at the ears of the listener sound signals, as if they actually originated at the ears, e.g., as though the listener were wearing earphones. This mode of operation has particular application to so-called artificial or quasi-stereo reproduction wherein crosstalk between the indivivdual channels must be minimized. Typically, different versions or modifications of a single monophonic signal are delivered independently to a listener's two ears to yield an effect which, ambiophony or spatial spread is similar to true stereophonic reproduction. By suitably adjusting the parameters of localization networks **55** and **56**, the signals radiated from loudspeakers **1** and **2** may thus be transformed as if originating at the listener's ears, i.e., the crosstalk component may be completely removed.

The localization apparatus of the invention may also be used to equalize stereophonic signals before recording so that they possess the necessary psychological directivity clues that will give rise to the phantom source illusion when the recording is subsequently reproduced with ordinary two-channel equipment. Apparatus of this sort is illustrated in FIG. 6. In the figure, a number of microphones **61–1** . . . **61–n** are employed in a studio **60** to capture sounds emanating from a variety of sources (not shown). The signals are independently amplified in amplifiers **62** and delivered to localization networks **63**. Each of the networks is adjusted for a particular angle ϕ associated with the corresponding microphone **61**. With the arrangement shown, therefore, the signals from the microphones, which may be spaced close to one another in one plane, may be made to appear to have originated from widely spaced sources. Even if the microphones themselves are spatially oriented, as, for example, in the production of a stereophonic recording, the illusion of separation may be further enhanced appreciably by passing the signals developed through localization networks **63**. The several signals $x_1(t)$ developed in the networks **63** are combined and supplied by way of amplifier **64** to output terminal **65**. Similarly, the $x_2(t)$ signals developed at the other output of each of the localization networks are combined and supplied by way of amplifier **66** to output terminal **67**. Normally, the signals which appear at terminals **65** and **67** are passed through the usual auxiliary equipment prior to recording on disc or tape. Since the recorded signals prepared in the manner described above contain the necessary directional information, reproduction of the signals on ordinary equipment will give rise to an apparent spread of the signals over a half plane area, even though only two loudspeakers are used. With either of the techniques described above, the stage width produced by two loudspeakers can be considerably increased.

The above-described arrangements are, of course, merely illustrative of the application of the principles of the invention. Numerous other arrangements may be devised by those skilled in the art without departing from the spirit and scope of the invention.

What is claimed is:

1. A two-channel sound system characterized by a broadened stage width comprising, first and second loudspeakers in spaced relation, localization network means supplied with signals from a first signal source for developing a sequence of delayed repetitions of said first signal, each replica having its magnitude adjusted according to a first schedule, localization network means supplied with signals from a second signal source for developing a sequence of delayed repetitions of said second signal, each repetition having its magnitude adjusted according to a second schedule, and means for supplying said repetitive signals from said localization networks to said first and said second loudspeakers, respectively, whereby the combined sound pressure field from said first and said second loudspeakers has prescribed phase and intensity characteristics at two different locations.

2. The sound system of claim **1** wherein said localization network means comprises: a first adding network supplied with sound signals from a source external to said network, and with periodic repetitions of said sound signals each adjusted in magnitude by a prescribed decrement; a first and second path for signals from said first adding means to first and second output terminals; the first of said paths including a second adding network, means for supplying signals from said first adding network directly to said second adding network, means for supplying signals delayed and attenuated according to a first prescribed schedule from said first adding network to said second adding network, and means for supplying the additive output of said second adding means to said first output terminal; the second of said paths including a third adding network, means for supplying signals delayed and attenuated according to a second prescribed schedule from said first adding network to said third adding network, means for supplying signals delayed and attenuated according to a third prescribed schedule from said first adding network to said third adding network, and means for supplying the additive output of said third adding means to said second output terminal.

3. A two-channel signal processing system comprising: first and second input terminals for receiving sound signals; first localization network means for developing first and second correlated signals, each including a plurality of variously delayed repetitions with prescribed intensity and phase characteristics, from signals supplied thereto from said first input terminal; second localization network means for developing first and second correlated signals, each including a plurality of variously delayed repetitions with prescribed intensity and phase characteristics, from signals supplied thereto from said second input terminal; first means for algebraically combining said first signals from said first and said second localization network means; second means for algebraically combining said second signals from said first and said second localization network means; means for delivering said algebraic combination of said first signals to a first output terminal; and means for delivering said algebraic combination of said second signals to a second output terminal.

4. A two-channel sound system comprising; first and second sources of sound signals, first and second loudspeakers in spaced relation; first localization network means for developing first and second correlated signals each including a plurality of periodically delayed repetitions with prescribed intensity and phase characteristics, from signals supplied thereto from said first source of sound signals; second localization network means for developing first and second correlated signals, each including a plurality of periodically delayed repetitions with prescribed intensity and phase characteristics, from signals supplied thereto from said second source of sound signals; first means for algebraically combining said first signals from said first and said second localization network means; second means for algebraically combining said second signals from said first and said second localization network means; means for delivering said algebraic combination of said first signals to said first loudspeaker; and means for delivering said algebraic combination of said second signals to said second loudspeaker.

5. A sound signal processing system comprising: a plurality of input terminals for the independent reception of sound signals; a plurality of localization networks, each supplied with signals from one of said input terminals, each of said localization networks including means for developing first and second trains of repetitive signals whose phase, intensity and frequency characteristics are related to each other to a prescribed degree; means for individually adjusting each of said localization networks to develop said first and said second correlated signals, respectively, according to a prescribed schedule, a different schedule being uniquely prescribed for each one of said localization networks; means for algebraically combining all of said first developed signals, means for algebraically combining all of said second developed signals; means for delivering said combination of first signals to a first output terminal; and means for delivering said combination of said second signals to a second output terminal.

6. Apparatus for creating arbitrarily located sound images comprising, a pair of loudspeakers, means for independently energizing each of said loudspeakers, localization network means associated with each of said energizing means for conforming the intensity, phase, and frequency characteristics of signals supplied thereto, in accordance with a prescribed schedule such that the resultant sound pressure waves radiated by both of said loudspeakers together combine at a location in spaced relation to said pair of loudspeakers to produce a resultant sound pressure wave whose subjective indicia indicate an origin other than in the area immediately encompassed by said pair of loudspeakers.

7. Apparatus for creating arbitrarily located sound images outside of the sector defined by a pair of loudspeakers and a point in front of and on the center line between them comprising a pair of correlated sound signals, means for independently developing from each of said signals a sequence of selectively attenuated and delayed repetitions of said signals, means for algebraically combining said first and second sequences according to a prescribed schedule to produce first and second driving signals, a pair of loudspeakers spaced apart from one another, and means for energizing each of said loudspeakers with one of said driving signals.

8. Apparatus for creating arbitrarily located sound images comprising first and second loudspeakers spaced apart from one another, a source of first and second

correlated signals whose relative phases and amplitudes specify a distinct spatial location of sound origin with regard to a pair of fixed points, means for iteratively producing from said first signal a number of signal replicas spaced apart in time and individually adjusted in magnitude in accordance with a first prescribed schedule, means for iteratively producing from said second signal a number of signal replicas spaced apart in time and individually adjusted in magnitude in accordance with a second prescribed schedule, said first and said second schedules being selected with relation to one another in accordance with the spatial relation of said first and said second loudspeakers and with relation to the sound pressure wave response desired at said spatial location of sound origin, means for supplying said first signal and its adjusted replicas to said first loudspeaker, and means for supplying said second signal and its adjusted replicas to said second loudspeaker.

9. In a two-channel sound system an input terminal, means for supplying sound signals to said input terminal, means for developing from said applied signals a sequence of replicas thereof each delayed by a prescribed interval and each adjusted in intensity and phase in accordance with a first prescribed schedule, a first output terminal, means for delivering said developed signals adjusted according to said first schedule to said first output terminal, means for developing from said applied signals a sequence of replicas thereof each delayed by a prescribed interval and each adjusted in intensity and phase in accordance with a second prescribed schedule, a second output terminal, and means for delivering said developed signals adjusted according to said second schedule to said second output terminal, said first and said second schedules being selected to yield signals at said first and said second output terminals with a desired degree of phase and intensity correlation.

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