

May 11, 1965

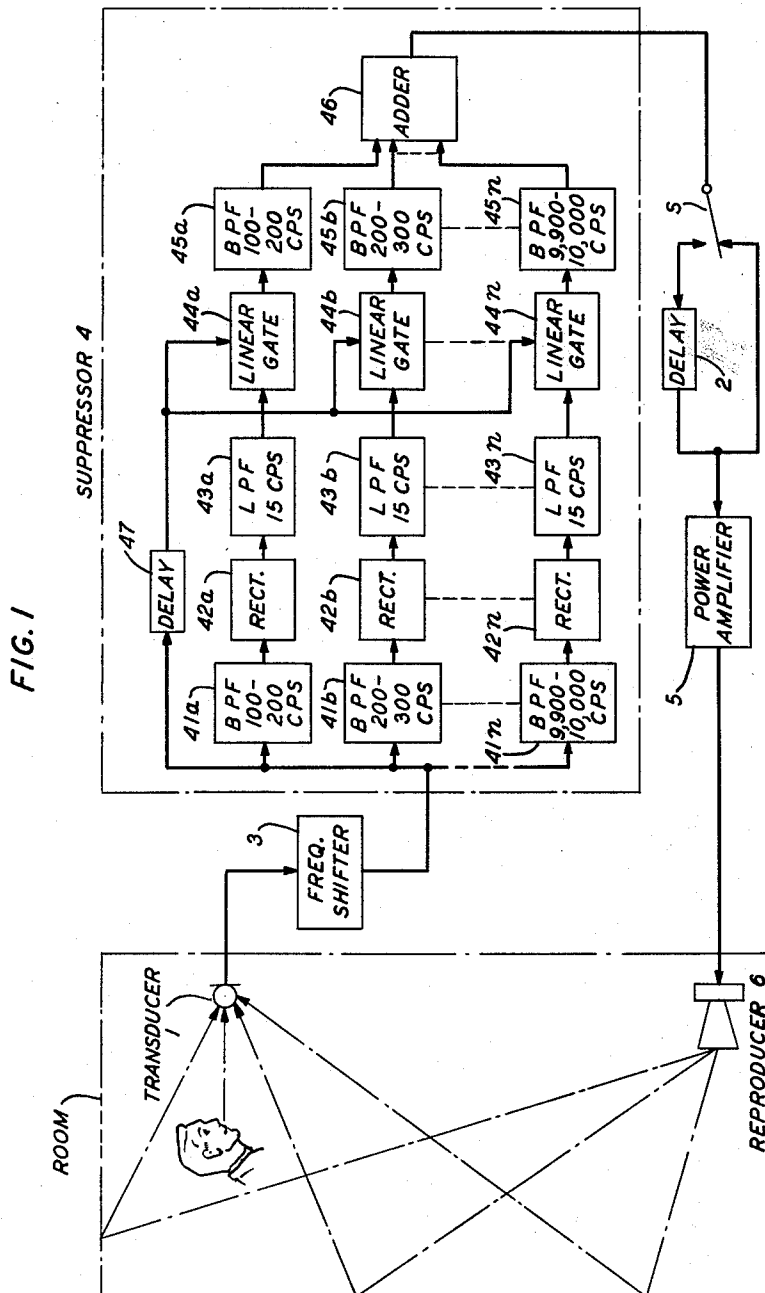
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3,183,304

SOUND AMPLIFICATION SYSTEM

Filed March 7, 1962

2 Sheets-Sheet 1



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FIG. 2A

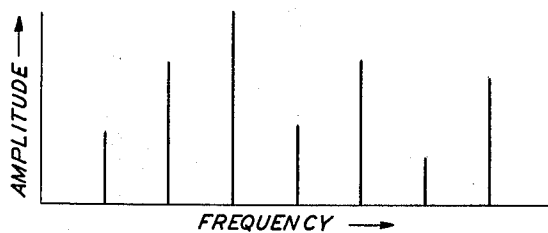


FIG. 2B

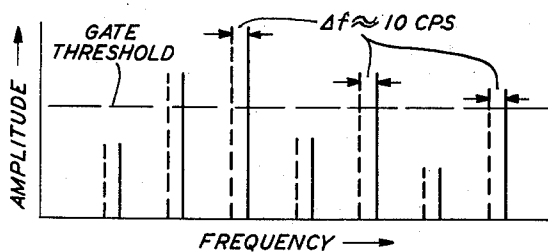


FIG. 2C

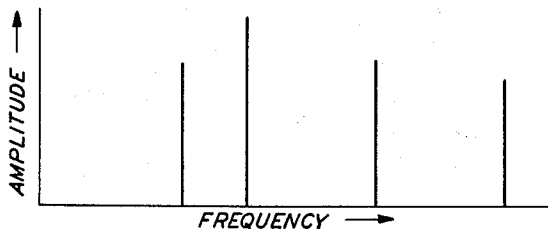


FIG. 2D

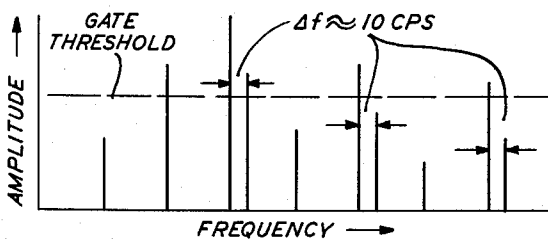
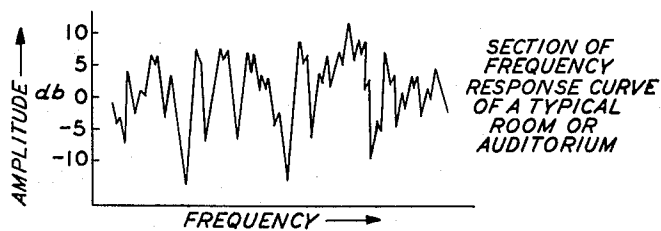


FIG. 3



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SOUND AMPLIFICATION SYSTEM

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5 Claims. (Cl. 179-1)

This invention relates to sound amplification systems, and in particular to sound amplification systems whose loudspeakers and microphones are located in the same sound field.

For a sound amplification system having a microphone and a loudspeaker located in the same sound field, for example, a public address system in an auditorium or a distant talking telephone system in an office, the well-known phenomenon of acoustic feedback between loudspeaker and microphone ordinarily prevents full use of the amplification of which the system is capable. Acoustic feedback occurs whenever an amplified sound emitted by a loudspeaker reaches the microphone, thereby establishing a feedback loop comprising the sound amplification system and the room. However, instability of the sound amplification system due to acoustic feedback occurs only when the loudness of the feedback sound reaching the microphone cumulatively increases on each trip around the feedback loop. In the absence of preventive measures, the cumulative increase in loudness results first in an audible "singing" noise, and finally in instability and a breakdown of the system, which is evidenced by a "howling" noise that obliterates the sound that it is desired to amplify.

Of the various types of feedback sound that cause instability in sound systems, one of the most universal is reverberant sound, which is caused by reflection from the room back to the microphone of sound emitted by a loudspeaker. Reverberant sound presents a universal problem because of the inherent nature of the so-called frequency response curves of ordinary rooms, the frequency response curve of an ordinary room being characterized by an irregular succession of several thousand closely spaced peaks and valleys within the frequency range of audible sound. For a sound amplification system operating at a sufficiently high gain level in such a room, there are generally one or more components of the emitted sound which coincide with peaks in the response curve, thereby sustaining the loudness of these components on reflection back to the microphone. On each trip around the feedback loop, the loudness of these components cumulatively increases, ultimately resulting in acoustic feedback instability.

One well-known method of preventing acoustic feedback instability caused by reverberant sound is to shift the frequency of each component passing through the sound system by a small, nearly imperceptible amount, on the order of two to five cycles per second. This frequency shift moves each reverberant component to a different point on the frequency response curve of the room on each trip around the feedback loop, thus making it highly probable that after several trips around the loop each reverberant component will have coincided with or fallen near a valley in the frequency response curve which will attenuate rather than sustain the loudness of each reverberant component. However, this method does not permit full use to be made of the amplification capabilities of a sound system because in the short but significant interval of time which is required for the frequency shift to become effective, the loudness of some reverberant components is cumulatively increased by an amount sufficient to cause a subjectively unpleasant singing noise. To prevent this singing noise, the gain of the system must be maintained at a level substantially below that which is theoretically attainable with frequency shifting.

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The present invention provides another solution to the problem of acoustic feedback instability, which permits a sound amplification system to be operated in an ordinary room or auditorium at a gain very near the theoretically attainable maximum without risk of instability. In this invention, there is inserted between the microphone and the power amplifier of a sound system a frequency shifter followed by a so-called "suppressor." The frequency shifter changes the frequency of each incoming sound component by a relatively small amount, and the frequency shifted components are applied to the suppressor. In the suppressor, the incoming components are separated from each other on the basis of frequency, and only those components whose amplitudes exceed a predetermined threshold are allowed to pass through the suppressor to the power amplifier. By appropriately adjusting the threshold, only components having relatively large amplitudes pass to the amplifier and all other components are suppressed. Since the components of direct sound, that is, sound received directly from a desired source such as a speaker, are characterized by amplitudes which are large relative to the amplitudes of reverberant sound, the suppressor operates to eliminate the reverberant components, thereby preventing acoustic feedback instability.

The frequency shifter and the suppressor of the present invention are effective in preventing acoustic feedback instability in each of the three situations that are possible when reverberant sound is detected by the microphone of a sound amplification system: first, the reverberant sound coincides in time with a later portion of the same direct sound from which the reverberant sound is derived; second, the reverberant sound coincides in time with a direct sound other than the direct sound from which the reverberant sound is derived; and third, the reverberant sound is not coincident in time with any other sound.

In the first and most difficult situation, the frequency shifter enables the suppressor to separate reverberant and direct sound components, despite the fact that the reverberant and the direct components originate from the same sound. The reverberant components differ in frequency from the direct components because of the change in frequency introduced by the frequency shifter during the passage of the earlier part of the direct sound through the system, the reverberant sound being derived from the earlier part of the direct sound. In addition, because the reverberant components are generally smaller in amplitude than the corresponding direct components, owing to attenuation by the air and reflection from the walls of the room en route to the microphone, the suppressor, with its appropriately adjusted threshold, eliminates the smaller reverberant components and passes the larger direct components. Some direct components of small amplitude are also eliminated by the suppressor, but most of the information content of sound is conveyed by the largest components, hence intelligibility is not impaired through the loss of these small amplitude direct components.

Although the suppressor is able to distinguish between components that lie relatively close to one another on the frequency scale, occasionally in the first situation a reverberant component and its corresponding direct component are not separated on the first trip of the reverberant component around the feedback loop, in which case the combined amplitude of the reverberant component and the corresponding direct component is generally large enough to exceed the suppressor threshold, thus enabling the reverberant component to pass through the suppressor together with the direct component to the power amplifier. However, on each successive trip around the feedback loop, the frequency shifter preceding the suppressor

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increases the frequency difference between a reverberant component and its corresponding direct component until after two or three trips, the suppressor is able to separate the two components and thereby eliminate the smaller reverberant component.

Further, each successive change in frequency introduced by the frequency shifter moves a reverberant component to a different point on the room frequency response curve, thereby increasing the probability that a reverberant component will coincide with a valley in the curve and be eliminated. The frequency shifter therefore performs two functions: to separate a smaller reverberant component from its corresponding larger direct component in order to enable the suppressor to eliminate the reverberant component; and to move a reverberant component to a different point on the frequency response curve of the room on each trip around the feedback loop in order to eliminate it by making it coincide with or fall near a valley in the response curve.

In the second and third situations, the present invention is even more effective in preventing acoustic feedback instability. When reverberant sound coincides in time with a direct sound other than the direct sound from which the reverberant sound is derived, it is only in random instances that a reverberant component lies sufficiently close on the frequency scale to a direct component for the suppressor to be unable to separate all reverberant components from all direct components; therefore, most reverberant components will be eliminated in the suppressor on their first trip around the feedback loop in the second situation. In the third situation, when the reverberant sound does not coincide with any direct sound, there are, of course, no direct sound components which would enable reverberant components to pass through the suppressor in the fashion described above; hence all reverberant components are eliminated by the suppressor on their first trip around the feedback loop.

The invention will be fully understood from the following detailed description of an illustrative embodiment thereof taken in connection with the appended drawings, in which:

FIG. 1 is a schematic block diagram showing a sound amplification system embodying the principles of this invention;

FIGS. 2A, 2B, 2C, and 2D are amplitude spectrum diagrams of assistance in explaining the apparatus of FIG. 1; and

FIG. 3 is a section of the frequency response curve of a typical room or auditorium.

Referring first to FIG. 1, there is shown a sound amplification system embodying the principles of this invention, in which both transducer 1 and reproducer 6 of the system are located in a typical room or auditorium characterized by a highly irregular frequency response curve of the type shown in FIG. 3. Any sound within the room that is detected by transducer 1, which may be a microphone of conventional design, is converted into an electrical wave, and this wave is delivered to frequency shifter 3.

Frequency shifter 3 is of well-known construction, and it operates to change the frequency of all the components of the wave by a uniform, predetermined amount, Δf cycles per second, where an appropriate value for Δf may be on the order of ten. The output terminal of frequency shifter 3 is connected to the input terminal of suppressor 4, and within suppressor 4, the incoming wave is applied in parallel to delay element 47 and to n identical subpaths, where the i th subpath comprises bandpass filter 41*i*, rectifier 42*i*, low-pass filter 43*i*, gate 44*i*, and bandpass filter 45*i*, all connected in series. Bandpass filters 41*a* through 41*n* are provided with relatively narrow, contiguous pass bands of uniform width which span the frequency range of the incoming wave. A suitable width may lie in the range between thirty and three hundred cycles per second; for example, with a width of one

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hundred cycles per second, an incoming wave whose frequency range extends from one hundred to ten thousand cycles per second requires

$$n = \frac{9900}{100} = 99$$

subpaths, as indicated in FIG. 1. Filters 41*a* through 41*n* separate the incoming wave into its individual frequency components, and rectifiers 42*a* through 42*n* followed by low-pass filters 43*a* through 43*n* develop, in well-known fashion, a set of control signals representatives of the amplitudes of the frequency components passed by the preceding bandpass filters.

Since the amplitudes of the frequency components of a sound constitute the amplitude spectrum of the sound, the control signals appearing at the output terminals of low-pass filters 43*a* through 43*n* represent the amplitude spectrum of a sound. The amplitude spectrum of a typical voiced speech sound before passing through frequency shifter 3 is illustrated graphically in FIG. 2A, where the heights of the regularly spaced vertical lines indicate the amplitudes of the various frequency components that occur at harmonics of the so-called fundamental frequency, f_0 , of the sound. The same spectrum after passage through frequency shifter 3 is shown in FIG. 2B, where the broken vertical lines indicate the original components and the solid lines indicate the frequency shifted components.

Gates 44*a* through 44*n* following low-pass filters 43*a* through 43*n* are conventional linear gates of identical construction, and each gate is provided with a control terminal, an input terminal, and an output terminal. Each of these gates operates in well-known fashion so that when a sufficiently large control signal is impressed upon its control terminal, any signal simultaneously applied to its input terminal is immediately passed in substantially unaltered form to its output terminal.

In the present invention, the output terminals of low-pass filters 43*a* through 43*n* are connected to the control terminals of gates 44*a* through 44*n*, respectively, and the output terminal of delay element 47 is connected in parallel to all of the input terminals of gates 44*a* through 44*n*. By suitably adjusting the bias of each of the gates 44*a* through 44*n* to respond only to those control signals whose magnitudes exceed a sufficiently high, predetermined threshold, the incoming wave from delay element 47 will appear only at the output terminals of those gates that receive control signals representing frequency components which occur at peaks or formants of the amplitude spectrum of the incoming wave. Similarly, since bandpass filters 45*a* through 45*n*, which are connected to the output terminals of gates 44*a* through 44*n*, respectively, have pass bands that are identical with the pass bands of corresponding bandpass filters 41*a* through 41*n*, the signals appearing at the output terminals of filters 45*a* through 45*n* will be the frequency components that occur at the formants of the amplitude spectrum of the incoming wave from delay element 47.

It is well known that the frequency components which occur at spectral formants of a sound convey most of the information content of the sound, hence the elimination of all but the largest frequency components by suppressor 4 does not impair the intelligibility of the sound. The components passed by filters 45*a* through 45*n* are combined in a conventional adder 46 to form a reconstructed wave which is amplified in power amplifier 5 and reproduced as audible sound in the room by reproducer 6, which may be a conventional loudspeaker.

If desired, the reconstructed wave from adder 6 may be passed to amplifier 5 through delay element 2 instead of directly by setting switch S to the appropriate position. In this way, a delay interval additional to that created by passage through suppressor 4 may be introduced, this additional delay serving to separate still further in time reverberant sound from direct sound. This delay helps

to synchronize the reception of a sound by distant listeners in a large auditorium with its utterance by a speaker, and in the case of extended or sustained sounds, it also helps to reduce the likelihood that reverberant sound will coincide at the microphone with the same direct sound from which it originated, thereby improving the effectiveness of this invention in preventing acoustic feedback instability. A suitable delay interval may be on the order of ten milliseconds.

The relationship of a suitable gate threshold to the amplitudes of the frequency components of a typical sound is illustrated by the horizontal broken line in FIG. 2B, where it is observed that the amplitudes of only four components exceed the threshold. The spectrum of the wave reconstructed by adder 46 from these four components is shown in FIG. 2C. It is apparent from FIGS. 2B and 2C that the number of components appearing in the spectrum of the reconstructed wave may be increased or decreased by appropriately adjusting the gate threshold, the number of components shown in FIG. 2C being merely illustrative of the principles of this invention.

The sound reproduced by loudspeaker 6 from the reconstructed wave and emitted in an ordinary room is typically reflected back to microphone 1 over various feedback paths, thereby establishing a feedback loop within the sound amplification system and the room. Some of the possible feedback paths in the room shown in FIG. 1 are indicated by broken lines connecting loudspeaker 6 with microphone 1.

However, not all of the components of the sound emitted by loudspeaker 6 appear in the reverberant sound reaching microphone 1 because those components of the emitted sound that coincide in frequency with substantial valleys in the frequency response curve of the room are too attenuated by reflection to be detected by microphone 1. In addition, even those components that coincide with peaks in the frequency response curve and are therefore sufficiently strong to be detected by microphone 1 are generally smaller in amplitude than when they were emitted by loudspeaker 6, owing to losses on transmission through the air and on reflection from the walls of the room. Thus, the components of reverberant sound arriving at microphone 1 coincide in frequency but not in amplitude with components of sound emitted by loudspeaker 6, and some components of emitted sound do not have any counterparts in reverberant sound. In addition, it is to be noted that because of the change in frequency introduced by frequency shifter 3, the reverberant components do not coincide in frequency with components of the original direct sound but are shifted on the frequency scale by an amount equal to Δf cycles per second.

In discussing the operation of the present invention in preventing acoustic feedback instability due to reverberant sound, there are three possible situations of interest when reverberant sound reaches microphone 1: in the first situation, the original direct sound from which the reverberant sound is derived is still being produced by the source, as in the case of a sustained vowel or a musical sound; in the second situation, the original direct sound from which the reverberant sound is derived has stopped and has been replaced by some other direct sound; and in the third situation, no direct sound is being produced.

In the first situation, the voiced sound whose spectrum is shown in FIG. 2A will be taken as an example of an original direct sound that is still being produced by the source at the time that the reverberant sound arrives at microphone 1. As previously noted, the components of the reverberant sound coincide in frequency with certain components of the emitted sound from loudspeaker 6, but the components of the reverberant sound differ in frequency by Δf cycles per second from the corresponding components of the direct sound arriving at microphone 1, owing to the change in frequency introduced by frequency shifter 3 during the passage of the first part of the original sound through the system of this invention. As illustrated

in FIG. 2D, the amplitude spectrum of the combined direct and reverberant sound arriving at microphone 1 thus includes both the direct components shown in FIG. 2A and the reverberant components, where the reverberant components are indicated by smaller vertical lines that are displaced to the right on the frequency scale by an amount Δf from corresponding larger vertical lines representing direct components. It is observed in a comparison of FIGS. 2C and 2D that the first component of the emitted sound spectrum in FIG. 2C is missing from the reverberant sound components in the combined direct and reverberant sound spectrum in FIG. 2D, resulting from coincidence with a valley in the frequency response curve of the room. In addition, it is observed in FIG. 2D that the amplitudes of the last two reverberant components fall below the gate threshold because of losses in transmission over the feedback paths between loudspeaker 6 and microphone 1.

After microphone 1 converts the combined sound into an electrical wave, frequency shifter 3 changes the frequencies of both the direct components and the reverberant components of the wave by an equal amount but the relative displacement in frequency between direct and reverberant components remains unchanged. From frequency shifter 3, the frequency shifted wave is delivered to suppressor 4, where the wave is separated into its individual components by filters 41a through 41n. Because of the difference in frequency between the reverberant and direct components, it is highly probable that a reverberant component and its corresponding original component will lie within the pass bands of different band-pass filters, if not on the first trip of the reverberant components around the feedback loop, then on a succeeding trip. For example, if the contiguous pass bands of filters 41a through 41n are of a uniform thirty cycles per second width, and the frequency shift introduced by frequency shifter 3 is ten cycles per second, then at most three trips around the feedback loop would suffice to cause a reverberant component and its corresponding direct component to lie within the pass bands of different filters.

When a reverberant component does not lie in the same pass band as the corresponding direct component, it is apparent that the spectrum of the reconstructed wave formed by adder 46 will contain the reverberant component only when its amplitude exceeds the gate threshold. Because of the transmission losses previously mentioned, it is a relatively rare occurrence for the amplitude of a reverberant component to exceed a suitably high gate threshold. Equally important, even if a reverberant component does have sufficient amplitude to appear in the spectrum of the reconstructed wave, the change in frequency introduced by frequency shifter 3 moves such a component to a different point on the room frequency response curve, thereby making it very likely that the reverberant component will coincide with or fall near a valley in the response curve and be eliminated. Frequency shifter 3 thus operates to eliminate reverberant sound as a source of acoustic feedback instability in two ways: by separating each reverberant component from its corresponding direct component to facilitate suppression of individual reverberant components in suppressor 4, and by moving each reverberant component to a different point on the room frequency response curve to make the reverberant components coincide with valleys in the room response curve. It is clear that with each trip around the feedback loop, the probability of eliminating reverberant components increases as the change in frequency introduced by shifter 3 cumulatively increases.

In the event that a reverberant component does lie in the same pass band as its corresponding direct component, the resulting control signal appearing at the output terminal of the following low-pass filter represents the sum of the amplitudes of the two components, and therefore it is very probable that both the direct and the reverberant

components will appear in the spectrum of the reconstructed wave. But one or two subsequent trips around the feedback loop will ordinarily suffice to eliminate such reverberant components, either by coincidence with a valley in the room frequency response curve, or by suppression in suppressor 4, or by a combination of coincidence and suppression.

In the second and third situations, where reverberant sound arrives at microphone 1 either in time coincidence with a direct sound different from the sound from which the reverberant sound is derived, or in the absence of direct sound, respectively, the present invention operates even more effectively to prevent acoustic feedback instability. For the case in which the reverberant sound coincides in time with a different direct sound, it is highly improbable that a reverberant component will lie in the pass band of a bandpass filter 41a through 41n which also contains a direct sound component. Thus in the second situation, it is highly probable that reverberant components will be suppressed in suppressor 4 on a first trip around the feedback loop. Those reverberant components which by chance lie in the same pass band as a direct component will eventually be eliminated in the course of one or two subsequent trips around the feedback loop by the same mechanisms as in the first situation.

It is clear that in the third situation there are no direct components to aid reverberant components in passing through suppressor 4, and therefore only in the comparatively rare instance when the amplitude of a reverberant component exceeds the gate threshold will a reverberant component appear in the spectrum of the reconstructed wave. In the event that a reverberant component does pass through suppressor 4, it is subject to the same processes of elimination on subsequent trips around the feedback loop as reverberant components in the two preceding situations.

It is to be understood that the above-described arrangements are merely illustrative of applications 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. In a sound amplification system comprising a microphone, a power amplifier, and a loudspeaker, means in circuit relation with said microphone, amplifier and loudspeaker for preventing instability due to acoustic feedback between said loudspeaker and said microphone including
 - a frequency shifter for changing the frequency of each frequency component in the amplitude spectrum of an incoming wave by a uniform, predetermined amount, to form a frequency shifted wave having an amplitude spectrum with frequency components that differ from the corresponding frequency components of the spectrum of said incoming wave by said predetermined amount of frequency change, and means connected to said frequency shifter for individually suppressing in the spectrum of said frequency shifted wave each frequency component whose amplitude is less than a preassigned threshold.
2. A sound amplification system comprising
 - a transducer located in a sound field for converting audible sound into an electrical wave,
 - means for applying said electrical wave to a frequency shifter for changing the frequency of each component of an incoming electrical wave by a predetermined, constant amount to form a frequency shifted wave,
 - means for delivering said frequency shifted wave to a suppressing means, said suppressing means including
 - means for separating said frequency shifted wave into its individual components,
 - means for obtaining from each of said components a control signal representative of the amplitude of each component,

gating means responsive to said control signals and supplied with said frequency shifted wave for passing only those components of said wave whose amplitudes exceed a preassigned threshold, and means for reconstructing an electrical wave from the components passed by said gating means, means connected to said suppressing means for delaying said reconstructed wave for a selected time interval,

amplifying means connected to said delaying means for increasing the power of said delayed reconstructed electrical wave to form an amplified electrical wave, and

reproducing means located in the same sound field as said transducer means for converting said amplified electrical wave into audible sound.

3. Apparatus for preventing instability due to acoustic feedback in a sound amplification system including a microphone, a power amplifier, and a loudspeaker which comprises

frequency shifting means for changing the frequency of each frequency component in the amplitude spectrum of an incoming electrical wave from said microphone by a uniform, preassigned amount to develop a frequency shifted wave having an amplitude spectrum in which each frequency component differs from the corresponding frequency component in the spectrum of said incoming electrical wave by said preassigned amount of frequency change, and

means coacting with said frequency shifting means for individually suppressing in the spectrum of said frequency shifted wave each frequency component whose amplitude is less than a predetermined threshold and for individually passing unaltered to said power amplifier each frequency component in said frequency shifted wave spectrum whose amplitude is greater than said threshold.

4. A sound amplification system comprising a transducer located in a sound field for converting audible sound into an electrical wave,

means connected to said transducer for shifting the frequency of each component of an incoming electrical wave by a uniform, preassigned amount to develop a frequency shifted wave, means for applying said frequency shifted wave to a suppressing means which includes

a plurality of subpaths, each of which is provided with an input terminal and an output terminal and each of which contains a bandpass filter having a relatively narrow pass band for separating said frequency shifted wave into its individual frequency components, a rectifier, and a low-pass filter connected in series,

a plurality of gating means in one-to-one correspondence with said plurality of subpaths, wherein each gating means is provided with an input terminal, an output terminal, and a control terminal having a uniform, predetermined threshold,

means for connecting the output terminal of each subpath with the control terminal of its corresponding gating means,

first delaying means provided with an input terminal and an output terminal,

means for delivering said incoming frequency shifted wave to the input terminal of each subpath and the input terminal of said first delaying means,

means for connecting the output terminal of said delaying means to the input terminal of each of said gating means, a plurality of bandpass filters each of which is provided with a pass band corresponding to the pass band of the bandpass filter in a corresponding one of said subpaths, an input terminal connected to the output terminal of a corresponding gating means, and an output terminal,

adding means provided with a plurality of input terminals in one-to-one correspondence with the output terminals of said plurality of bandpass filters, and an output terminal, and
 means for connecting the output terminals of said plurality of bandpass filters to the corresponding input terminals of said adding means,
 second delaying means provided with an input terminal and an output terminal,
 means for connecting the output terminal of said adding means to the input terminal of said second delaying means,
 power amplifier means connected to the output terminal of said second delaying means, and
 reproducing means following said power amplifier means for converting an electrical wave from said power amplifier means into audible sound.
 5. Apparatus for preventing instability in a sound amplification system due to acoustic feedback between a loudspeaker and a microphone of said system which comprises
 a microphone located in a sound field of both direct sound and reverberant sound for converting said direct sound and said reverberant sound into an electrical wave having an amplitude spectrum containing direct frequency components that correspond to the frequency components of said direct sound and reverberant frequency components that correspond to the frequency components of said reverberant sound,
 frequency shifting means supplied with said electrical wave for shifting each direct frequency component and each reverberant frequency component of said electrical wave by a uniform, predetermined amount of frequency to form a frequency shifted wave having an amplitude spectrum containing direct frequency components and reverberant frequency

quency components that differ by said predetermined amount of frequency from the corresponding direct frequency components and reverberant frequency components in the amplitude spectrum of said electrical wave,
 means connected to said frequency shifting means for suppressing substantially all of said reverberant frequency components in the amplitude spectrum of said frequency shifted wave, said suppressing means including
 means for separating the spectrum of said frequency shifted wave into its individual frequency components, and
 means responsive to said individual frequency components for individually suppressing in the spectrum of said frequency shifted wave each frequency component having an amplitude that is smaller than a preassigned threshold and for individually transmitting each frequency component in said frequency shifted wave spectrum which has an amplitude that exceeds said threshold to form the spectrum of a reconstructed wave,
 amplifier means supplied with said reconstructed wave for amplifying said reconstructed wave, and
 loudspeaker means connected to said amplifier means for reproducing audible sound from said amplified reconstructed wave.

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ROBERT H. ROSE, *Primary Examiner*.WILLIAM C. COOPER, *Examiner*.