March 30, 1943. L. A. DE ROSA 2,315,248

PSEUDO-EXTENSION OF FREQUENCY BANDS

Filed July 30, 1940 5 Sheets-Sheet 1

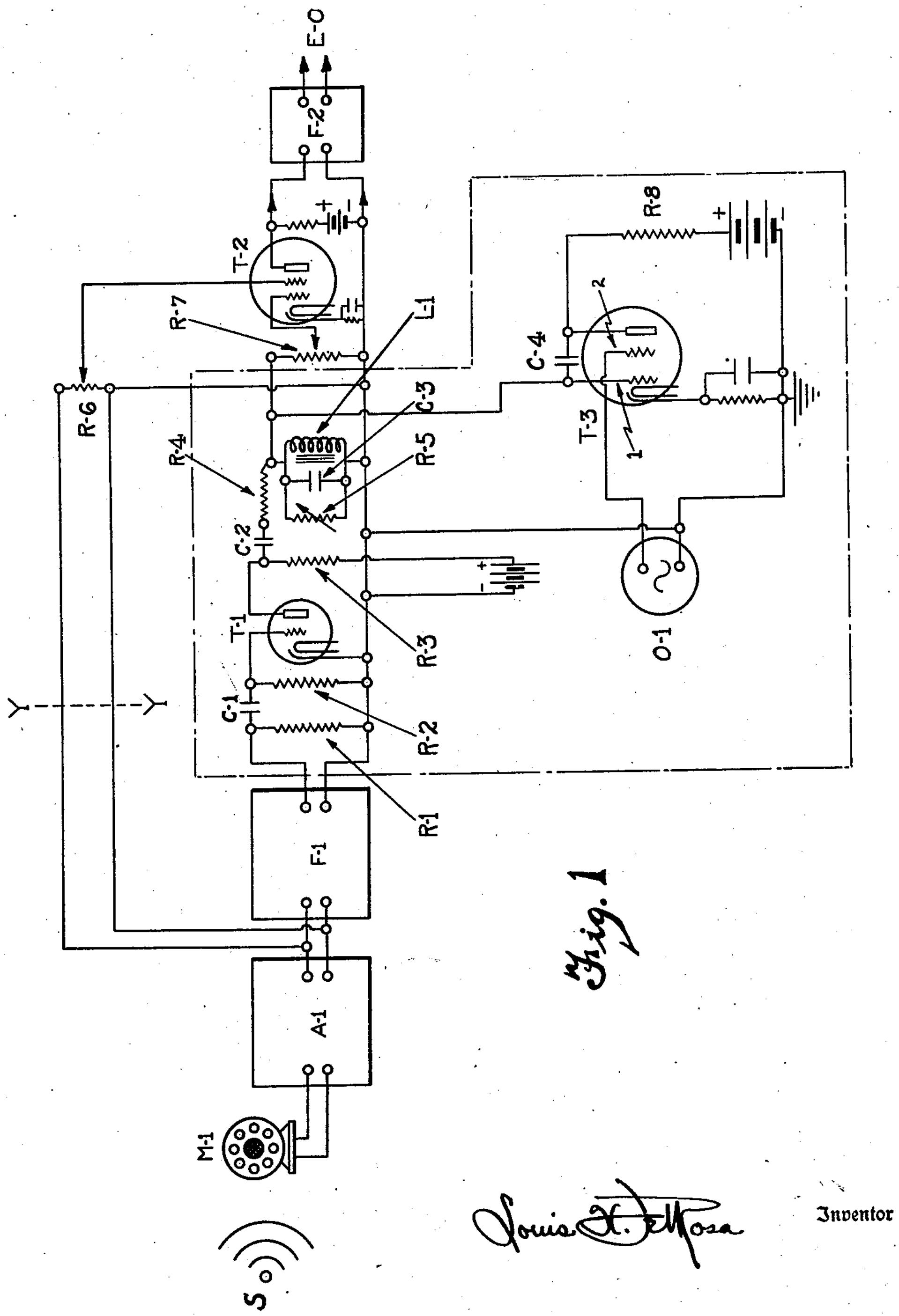

Fig. 1

Louis A. DeRosa

Inventor

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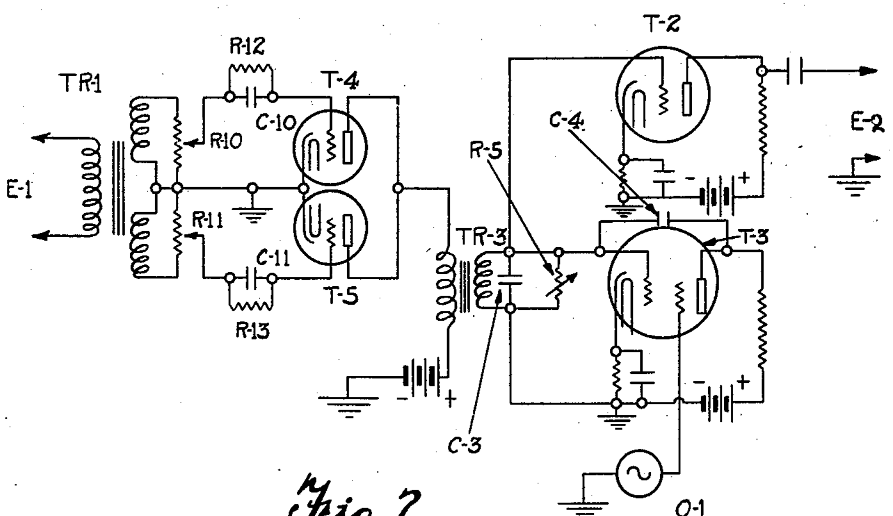

Fig. 2

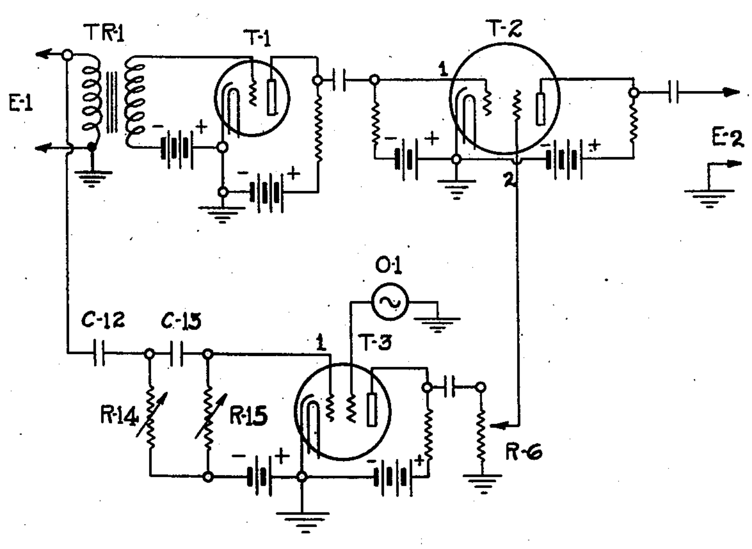

Fig. 3

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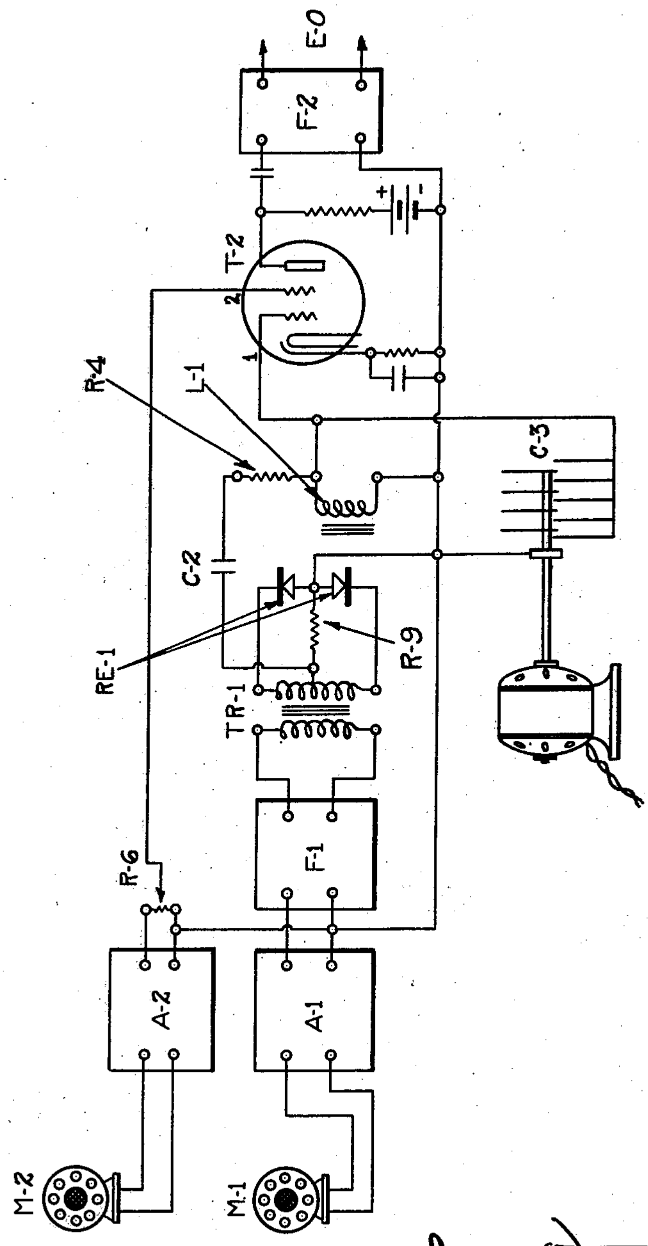

Fig. 4

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PSEUDO-EXTENSION OF FREQUENCY BANDS

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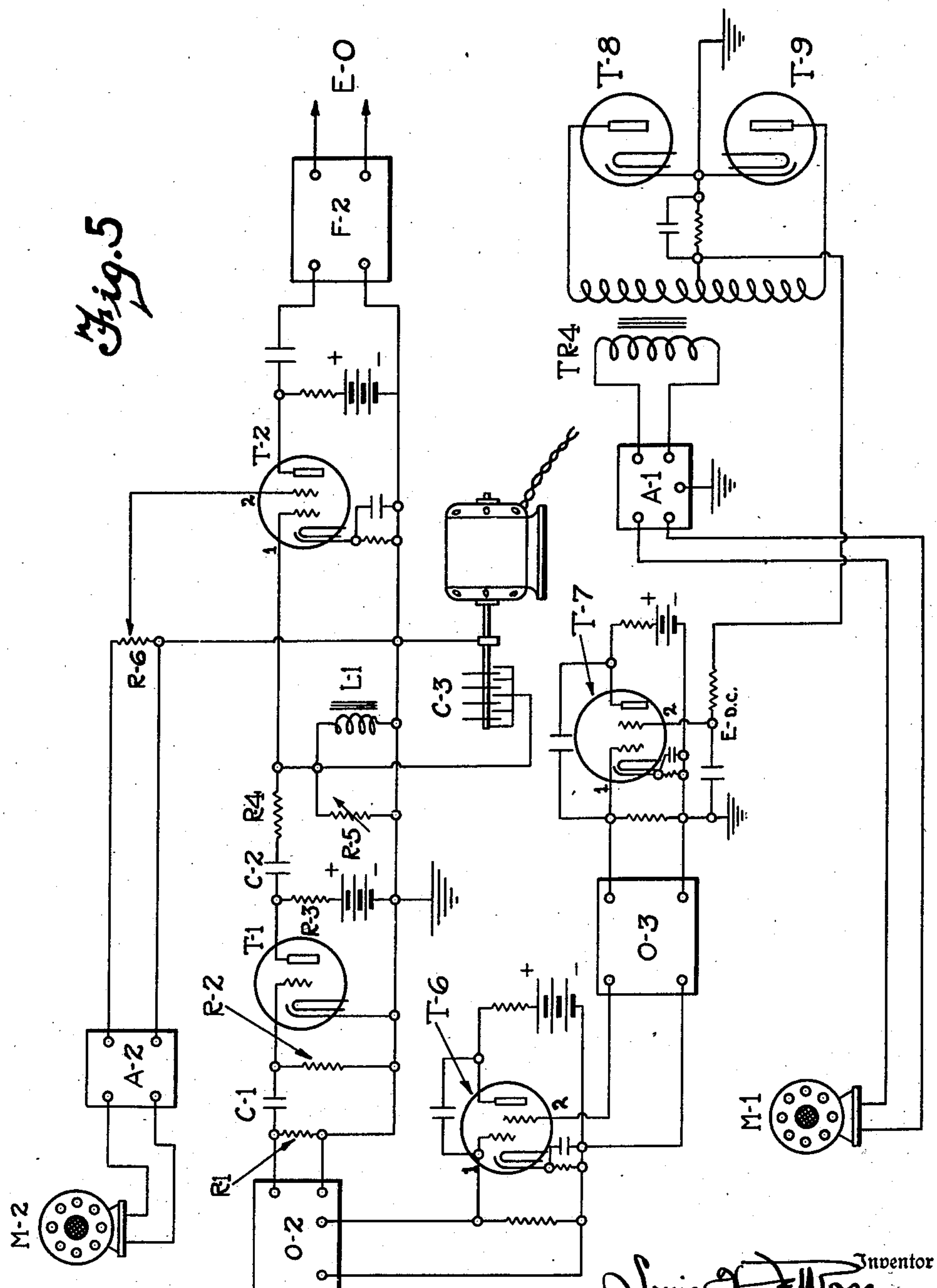
Fig. 5



Louis A. DeRosa
Inventor

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PSEUDO-EXTENSION OF FREQUENCY BANDS

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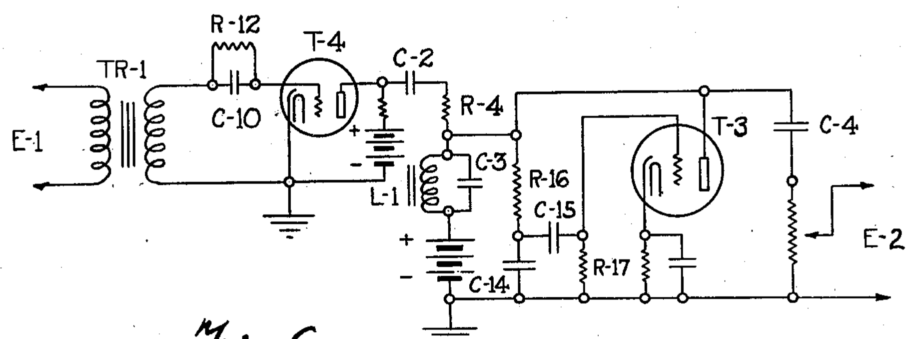

Fig. 6

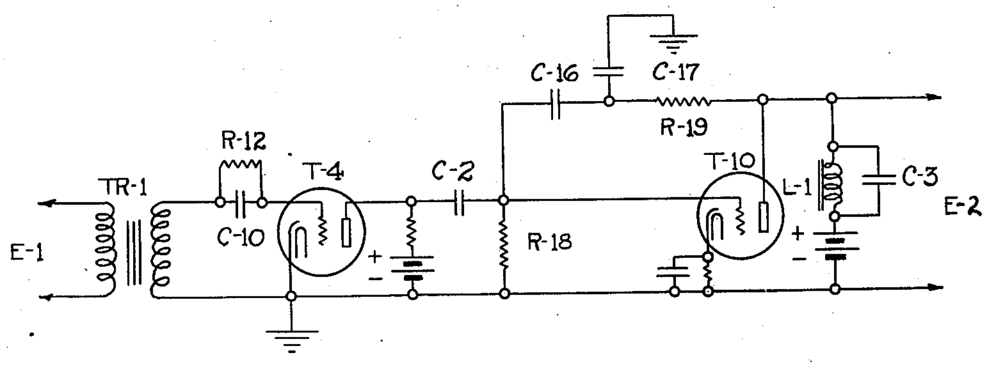

Fig. 7

Louis A. DeRosa Inventor

Patented Mar. 30, 1943 **2,315,248**

UNITED STATES PATENT OFFICE

2,315,248

PSEUDO-EXTENSION OF FREQUENCY BANDS

Louis A. de Rosa, Dayton, Ohio

Application July 30, 1940, Serial No. 348,359

21 Claims. (Cl. 179—1)

This invention deals with the pseudo-extension of frequency bands and particularly with improvements in the method and means wherein an audio signal, at some point or at some time in its transmission either directly or indirectly to the ear, is modified so that, while all the composite frequencies present in the original audio signal are not present in the signal ultimately transmitted to the ear, the auditory perception is of a sound which has substantially all the sonant characteristics of the original audio signal. This modification process has hitherto been termed, "Pseudo-extension of frequency bands" in the U. S. Patent to Bagno No. 2,113,976. This invention also deals with methods and means wherein a basic audio signal, during the course of transmission to the ear, is modified and improved so that its final effect on the ear may more nearly simulate a predetermined tonal characteristic.

One object of this invention is to improve the apparent faithfulness to the ear of sounds which have been transmitted to the ear thru a system having an acceptance band which is narrower than that required to transmit the composite frequencies of those sounds.

Another object of this invention is to afford a method whereby brilliance and richness may be imparted to an electrically transmitted sound.

Another object of this invention is to provide a method of retaining and even enhancing the percusston, clangor or transient qualities of a transmitted sound.

Another object of this invention is to furnish a method and means by which at least a portion of the sonant characteristics of a sound are transmitted by a system and so modified during transmission so as to produce an effect on the listener that an apparent larger portion of the sonic characteristics of the sound have been transmitted to him.

Another object of the invention is to provide an improved method of pseudo-extension whereby radio reception utilizing limited side-band transmission may be given better over-all fidelity by circuit modifications at the broadcast transmitter.

Another object of the invention is to afford an improved system of pseudo-extension whereby the present day methods of recording and reproducing sounds and intelligence faithfully are simplified and rendered less costly.

Another object of this invention is to provide an improved method of pseudo-extension whereby persons possessing certain types of auditory deficiencies are better enabled to hear sounds to which they are not normally responsive.

Other objects, functions and serviceable applicabilities will be apparent from the description and claims which follow.

Fig. 1 shows a diagrammatic circuit of one embodiment of the invention.

Fig. 2 shows, diagrammatically, an alternate circuit which may be employed to utilize the advantages of the invention.

Fig. 3 depicts a diagrammatical representation of another alternate of the apparatus of Fig. 2.

Fig. 4 shows, diagrammatically, an electromechanical alternate utilizing two microphones, one for bass and one for high frequency pick-up.

Fig. 5 illustrates an application of the invention for the purpose of creating or augmenting brilliance and richness, and for applying a controllable degree of said brilliance and richness to an audio channel.

Fig. 6 shows, diagrammatically, another possible arrangement for obtaining an improved pseudo-extension effect.

Fig. 7 illustrates another circuit embodying the principles of the invention.

Throughout the drawings, parts performing like functions have been given the same reference numbers.

Before giving a detailed description of the improvements involved in this invention, it is believed that a brief explanation of the action of the ear would facilitate the understanding of the disclosure.

The ear may be regarded as a delicate translating system which receives acoustical vibrations and converts these vibrations into nerve-impulses. These nerve-impulses are transmitted along the eighth cranial nerve to the superior temporal convolution of the cerebral cortex. If a single audible frequency is conveyed to the ear and the frequency of this acoustical stimulus is varied from a low value, of, for example 400 cycles per second to successively higher frequencies and at the same time a fibre of the eighth cranial nerve is examined by a coaxial prod, associated with an audio amplifier and an oscilloscope, it will be found that the nerve impulses vary in number with the stimulating frequency until that stimulating frequency is increased to a value of approximately 900 to 1000 cycles per second. Above this point, an impulse occurs at every second cycle of the stimulating frequency until the stimulating frequency is increased to about 3000 cycles per second at which time a nerve fibre contains an impulse on every third cycle of the stimulating signal. At the highest audio frequencies, the impulses do not conform in any way with the periodicity of the stimulating frequency but assume a random character. It seems evident, therefore, that the sensations accompanying the presence of higher frequency stimuli must be conveyed to the brain by a pattern contributed to by a large number of nerve cells, no individual cell being able to furnish impulses of frequency higher than probably 900 to 1000 cycles per second.

The exact process by which the high frequencies, probably those above 4000 cycles, are translated into sensations recognizable by the brain as being indicative of the presence of these high frequencies, is a highly hypothetical one. The suppositions advanced by various research workers in the field are numerous and contradictory. An hypothesis to explain the process would only be of academic interest and would not aid or clarify the method or means associated with this invention.

The improved process of pseudo-extension is, however, probably made possible by two effects occurring in the ear. First, to produce the sensation of hearing a certain frequency, it is not necessary that the stimuli must excite any definite section of the basilar membrane. Therefore, the alternation and successive stimulation that the nerve cells undergo when a high frequency is heard can be produced by a lower frequency which is transitionally or dynamically varied as a function of time so as to alternately and successively stimulate adjacent nerve terminations on the basilar membrane. Secondly, due to the non-linearity of the hearing mechanism. the intermodulation frequencies formed by the pseudo-extension process tend to produce high frequency subjective tones in the ear. As an illustrative experiment, this second effect can be made somewhat more noticeable by the use, in conjunction with the pseudo-extension circuits shown hereafter, of a loudspeaker which, under ordinary operating conditions, has a high percentage of distortion.

Besides the direct audible recognition of sustained high frequencies, there are other effects, such as percussion and clangor, associated with certain sounds, the transmission of which ordinarily necessitates a circuit or circuits capable of transmitting practically the entire audio spectrum. An explanation of this follows:

Every practical amplitude modulated medium, such as an audio signal, can be represented mathematically by a Fourier series. A trigonometric function would represent an infinitely long stimulus and is therefore seldom encountered in practical considerations. Both the onset and the decay of a sound are always accompanied by a transient respectively, to and from a relatively steady state. This onset or decay has associated with it an infinite number of sidebands. These transient conditions are co-existent both for periodic and non-periodic functions. For a transducer, such as the ear, having the equivalent of a zero steady pressure response, the ultimate or steady state response of a non-alternating or direct stimulus or signal is absent, but the onset and decay effects must be considered for fidelity transmission.

As the duration of the signal or sound stimulus is increased, the larger will be the proportion of its total energy that will be localized at the frequencies of a similar tone of infinite duration.

The composite frequency spectrum of the sound stimulus or signal, regarding it as being infinite, continuous and with only a finite number of discontinuities, will determine the action of a sudden onset or decay of that stimulus, insofar as the energy distribution in the associated frequency side-bands and their relative magnitudes.

An analysis by means of the Fourier Integral discloses the fact that the energy in the transient terms or side-bands becomes less and less as the rate of build-up or the rate of decay becomes slower and slower; simultaneously the mean energy of these transients or side-bands is shifted towards the lower frequencies. If the complexity of the steady state stimulus per se is great, the transient or side-band energies become correspondingly larger.

It is evident from the foregoing explanation, that although a sound or audio signal is of low frequency, if it is of short duration, the transmission circuit must, in present day practice, be capable of transmitting practically the entire audio spectrum in order for this sound or audio signal to sound realistic to the ear.

The action of a non-linear circuit element for a signal wave of transient nature is to increase the complexity and energy of the side-band components and therefore produce upon the ear, the effect of an augmented transient quality. As an example, the energy in the side-bands associated with an increase or decrease in amplitude of a signal depends on the rapidity with which the envelope of the signal builds up or decays respectively. If such a signal is applied to a non-linear circuit element, the resulting signal contains numerous "distortion" frequencies due to the intermodulation of the side-band frequencies and the steady state frequencies of the signal. The result therefore is to increase the proportionate energy in the side band frequencies and thus produce the effect of an augmented transient upon the listener.

By utilizing the improved process of pseudo-extension hereinafter described, it is possible to retain and even augment the percussion, clangor or transient qualities of a transmitted sound, although the transmission may be by means of a band considerably narrower than necessary to transmit the entire audio spectrum. The improved process of pseudo-extension produces also a sensation of auditory perspective similar to that effected by binaural transmission.

This invention, in its most elementary form consists of two main operating circuits, the first of these, hereafter called the "complexity" or "X" generator, is a circuit arranged to produce a complex audio spectrum. By a complex audio spectrum is meant a group of audio frequencies, the prime factor of which frequencies is a small number and may actually be smaller than unity. This "X" generator may be a device having a non-linear input vs. output characteristic such as a detector, may be a source of frequency modulated audio frequencies, phase modulated audio frequencies, or any other source of a complex audio spectrum.

To produce a pseudo-extended audio signal which has practically the full intelligence and sonic attributes of a full audio range signal it is necessary to create by means of the above mentioned "complexity" generators an extremely complex audio spectrum. If this multifrequency spectrum, consisting largely of frequencies not present in the original sound, is amplified and converted to acoustic power, the result is one of confusion and distortion to the ear, particularly during transient intervals. In my experience I have found that if the prior art circuits are adjusted carefully and the extraneous frequencies are limited in magnitude that a measure of "pseudo-extension" is possible. Attempts, in prior art circuits, to greatly increase the apparent frequency range of a narrow band "pseudo-extended" signal by increasing the complexity and number of cross-modulation or intermodulation frequencies results in a characteristic dissonance, harshness and roughness.

The second of the fundamental operating circuits is called a "consonator" or "C" circuit. The purpose of this "consonator" or "C" circuit is to adapt utilizable portions of the output of the "complexity" or "X" generator, and modify them so as to produce the subsequent effect of apparent brilliance and other desired auditory effects with a minimum of extraneous dissonance and minimum of what is generally termed distortion. In other words, it is the function of the "consonator" circuit, several forms of which are illustrated and described herein, to select, arrange and assort the complex audio spectrum so as to present to the ear an audio signal which, while containing numerous "extraneous" frequencies, will sound articulate, assonant and natural.

Since the advent of short-wave communication, I have observed an interesting phenomena associated with the transmission of audio signals by means of these high frequency radio waves. Audio signals, transmitted by short-wave carriers, and received on a recevier having a poor over-all fidelity characteristic, while at times badly distorted, had an apparent brilliance, richness and articulation greater than that which could be reasonably expected from such a receiver. It is my belief that, in those cases, the increase in the apparent frequency range of the transmitted audio signal was caused by the random, transitional phase-shifts and random, transitional frequency-selective transmission of the R. F. side-bands during propagation thru the ether medium. These random, transitional phase-shifts and random transitional frequency-selective occurrences produce an effect in the short-wave receiver, to a partial and very limited extent, similar to that produced by the "consonator" circuit of my invention.

Whereas the effect of this pseudo-extension phenomena associated with short-wave transmission has practically no value and is accompanied with severe distortion, my invention affords a method and means whereby the pseudo-extension, not a random, and accidental effect, is controllable in degree and magnitude; and being distortion-free, is entirely practical, utilizable, and of manifold applicability.

Besides these two main operating circuits; namely, the "X" and "C" circuits, there may be several auxiliary circuits, described hereinafter, such as input filters, control circuits, by-pass circuits and output filter circuits.

Referring to Fig. 1, S is a source of acoustical power, the signals of which are picked up by a microphone M—1 and amplified by amplifier A—1. The output of this amplifier (A—1) is impressed across the potentiometer R—6 and also across the input of an audio frequency selective network F—1. The signal impressed across R—6 is adjusted in amplitude to a convenient level, and impressed on one grid of the electronic mixer T—2 and in this manner combined with the products impressed on the other grid of this mixer tube T—2.

The other portion of the output of amplifier A—1 which is impressed on the audio frequency selective circuit F—1 is modified by F—1 so as to attenuate the middle and lower audio frequencies. Good results have been obtained without this filter F—1 but the brilliance may then need to be controlled by a pilot channel. In this case the pilot channel would be arranged to operate so as to increase the brilliance only at desired times. Practically, if F—1 is omitted, it is difficult to adjust the level of signal to the input of the "X" generator (in this case T—1), mainly because of the relatively large amplitudes of middle and low frequency components encountered in most music and speech. Using for F—1 a filter consisting of a single stage, constant K prototype, high pass with a cut-off frequency of about 1000 cycles has yielded good results. The Q of the inductances of the constant K prototype was about 4. The resistance R—1 is used to terminate the filter F—1 properly. The output of the filter F—1, is connected thru the grid leak and condenser combination of R—2 and C—1 respectively, to the X generator T—1. In this circuit arrangement (Fig. 1) the X generator is a grid leak detector with a plate load resistor R—3. Various types of non-linear "detectors," some of which are described later, can be used satisfactorily. In the circuit of Fig. 1, T—1 is a remote cut-off triode, if a remote cut-off triode is not available, a remote cut-off pentode, connected as a triode can be used. A remote cut-off tube is preferable so that the action of T—1 is not critically dependent on the amplitude of the signals impressed on its grid; in other words, the e_g—i_p characteristic, while non-linear, does not have an abrupt cut-off.

The "consonator" or "C" circuit, used in the circuit of Fig. 1 consists of a parallel tuned circuit C—3 and L—1, tube T—3 with its associated circuit, and a low frequency audio oscillator O—1.

Tube T—3 is arranged so that its transconductance (g_m) is varied at the audio oscillator frequency rate. This variation of g_m causes a corresponding change in the quadrature component of the tube input admittances. In this manner, utilizing the variation in input admittance of grid number 1 to ground, the effective capacity across the inductance L—1 is varied at the audio oscillator frequency; causing the resonance frequency of the tuned circuit to vary as a function of time, with an accompanying varying phase shift of the components accepted by the tuned circuit and transmitted. Good results are obtainable with O—1 set at about 7 cycles per second.

C—2 is a blocking condenser and R—4 is a high resistance used to increase the discrimination of the tuned circuit. R—5 is a variable resistance shunted across the tuned circuit so as to lower the sharpness of that circuit and allow transmission of a relatively wider portion of the complex frequency spectrum generated by T—1. This variable resistance R—5, when decreased in value, increases the clangor and percussive auditory effects for transient signals.

Those components of the output of T—1 selected by the "consonator" are then impressed on a grid of the mixer tube T—2 through the volume control R—7. The mixed output of tube T—2 is passed thru the low pass filter F—2 and transmitted to the utilization apparatus.

The low pass filter F—2 may be dispensed with if, in the subsequent transmission of the pseudo-extended signals, the high frequencies are, per se. lost or attenuated, or if it is desired to transmit the entire spectrum of the pseudo-extended signals.

The variable reactance tube T—3 should be adjusted, by any well-known means, so as to cause as little a change as possible in the conductance portion of its input admittance due to the oscillator (O—1) signal.

The quiescent frequency of the tuned circuit is selected below the cut-off frequency of the subsequent transmission apparatus. Good results have been obtained with this circuit, quiescently tuned to about 3500 cycles for a 4000 cycle cut-off for F—2. For a 3000 cycle cut-off for F—2, a quiescent tuning point of about 2700 cycles has been found satisfactory.

The drawing of Fig. 2 shows an alternate arrangement for the "complexity" generator and "consonator" circuit. The apparatus arrangement of Fig. 2 could be used in place of the arrangement of Fig. 1 shown in the dot-dash enclosure. The "X" generator, in this case, is a push-pull input, transformer coupled to two grid leak detectors T—4 and T—5. The outputs of these tubes are connected in parallel and transformer coupled to the "C" circuit. The advantage of this type of circuit is that the input frequencies, impressed at E—1, can be adjusted by means of R—10 and R—11 so as to limit their amplitudes in the output of T—4 and T—5 to the value of the higher order products generated by the nonlinear "X" generator tubes T—4 and T—5. This circuit is also of academic importance in demonstrating the operation of the "C" circuit when supplied with a pattern consisting entirely of extraneous or so called distortion frequencies. Transformer TR—3 is chosen to have a high leakage reactance and this leakage reactance is used, in conjunction with C—3 and the varying reactance component of the input admittance of T—3, to supply the tuned selector circuit corresponding operationally to L—1, C—3 and the T—3 arrangement of Fig. 1. The resistor R—5 in Fig. 2 plays the same role as its counterpart R—5 of Fig. 1. The output of Fig. 2, E—2 may be mixed with a low frequency channel, passed thru a low pass filter and then advantageously utilized.

Fig. 3 shows another alternate arrangement for the apparatus in the dot-dash enclosure of Fig. 1. The circuit of Fig. 3 illustrates a set-up utilizing the principles of this invention wherein the higher audio frequencies E—1 are divided into two channels. The top portion is applied, thru an audio transformer TR—1 to an over-biased vacuum tube T—1. The output of this tube is fed to one grid of a twin grid electronic mixer tube T—2. The lower portion of the divided output of E—1 is impressed on a phase shifting net-work C—12, R—14, C—13, R—15. The output of this phase shifting network is applied to the number 1 grid of a twin grid amplifier tube T—3. On the other grid of T—3, the output of a low frequency audio oscillator O—1, adjusted roughly to 7 cycles per second, is applied. This oscillator, the setting of which is not critical, causes a low frequency amplitude modulation of the output of the phase-shifting network. The output of T—3 is applied thru a small capacity coupling condenser and a potentiometer R—6 to the number 2 grid of T—2, the fundamental and the neighboring low frequency side-bands due to O—1 are attenuated so as not to be annoying.

The potentiometer R—6, and the resistors R—14 and R—15 are adjusted so that when the oscillator signal from O—1 passes thru zero of its cycle, the signal on grid number 2 of tube T—2 is 180 degrees out of phase with the signal on grid number 1 and of a magnitude so as to make the plate signal approximately zero. As the oscillator signal goes thru its complete cycle the output of T—2 will be phase modulated and, incidentally, amplitude modulated, with the oscillator O—1 signal. The extent of phase modulation may be varied without serious disadvantage, in fact, for certain types of signal, circuit arrangements producing a larger phase shift than obtainable with the circuit of Fig. 3 have yielded better results.

The output of T—2, designated as E—2 may then be connected to a circuit where the bass channel such as Y—Y of Fig. 1 may be mixed and, after optional filtering thru a filter such as F—2, be transmitted for gainful service.

Fig. 4 shows a sound source S with two microphone pick-ups, M—2 being a microphone especially suitable for bass frequency pick-up while M—1 is a microphone suitable for high frequency pick-up. The output of M—2 is amplified by the amplifier A—2, the output of which is adjusted to a convenient level by the volume control R—6 and then impressed on the number 2 grid of an electronic mixer tube T—2.

The output of the microphone M—1 is amplified by the amplifier A—1, passed thru the selective filter F—1 and then impressed on another version of an "X" circuit, a full wave rectifying circuit consisting of two rectifiers RE—1. The rectified output is then impressed, thru an isolating resistor R—4 on the "C" or "consonator" circuit. The "consonator" circuit in this case consists of a motor driven variable condenser C—3 and the inductance L—1. The condenser C—3 is varied at a low audio rate of about 7 cycles per second and thus the phase and frequencies accepted by the tuned circuit are transitionally shifted. The signals thus appearing across C—3, L—1 are impressed thereafter on the number 1 grid of the electronic mixer tube T—2. The combined output of T—2 is subsequently, after passing thru the optional low pass filter F—2, transmitted to various optional transducers and converted finally to acoustic power to obtain the required auditory effect.

The circuit of Fig. 5 illustrates an application of the principles of this invention to the production of brilliance and richness to a degree, if desired, greater than that available in the primary acoustical source.

M—2 is a microphone arranged to pick up the middle and lower audio registers of an acoustical sound source. A microphone or a microphone with an associated filter arranged so that its output begins to drop off at about 1000 cycles and is approximately 10 db. down at 3000 cycles has been found satisfactory for M—2. The output of M—2 is amplified by amplifier A—2, adjusted in level by means of the control R—6 and then applied to the number 2 grid of an electronic mixer tube T—2.

The object of the second channel is to provide the "consonator" circuit, in this case, R—5, L—1 and C—3, with a complex pattern formed by the frequency modulation of approximately a 3500 cycle audio frequency by a low audio frequency. The sidebands of the 3500 cycle signal are spaced apart from the carrier by an interval equal to the modulating frequency, that is, the low audio frequency. The frequency of the low audio modulating frequency is varied depending on the intensity of high frequencies in the acoustical source, being lower for greater high frequency amplitudes and higher for smaller amplitudes of the high frequencies. When the acoustical source is emanating extreme brilliance, the modulating frequency should be approximately 7 cycles per second.

The microphone M—1 is arranged either by design or by some associated filter means, to have a high pass characteristic, with a drooping, limited acceptance for frequencies below approximately 3000 cycles. The selective output of the microphone M—1 is amplified by the audio amplifier A—1. The output of A—1 is transformer coupled to a full-wave audio rectifier circuit consisting of the two diodes T—8 and T—9. The D. C. component of the rectified output of these diodes is filtered and applied to the number 2 grid of a two grid tube T—7.

The tube T—7 is connected so that any variation in the D. C. voltage applied to its number 2 grid will cause a change in its transconductance and hence a change in the magnitude of the susceptance or quadrature component of the input admittance between grid number 1 and ground. This input admittance is shunted across one of the beat frequency oscillator circuits of O—3 so that, when the susceptance between grid number 1 and ground is varied, the resulting audio beat frequency is varied. This resulting audio beat frequency is, by its effect on the transconductance of T—6, caused to vary the input susceptance of grid number 1 to ground at the audio beat frequency rate.

The input susceptance of grid number 1 of T—6 is shunted across the tuned circuit of one of the beating oscillators of the beat note oscillator O—2 so as to frequency modulate the output of O—2.

To further increase the complexity of the sideband pattern thus produced; the output of beat note oscillator O—2 is impressed on a grid leak detector T—1 thru the grid condenser C—1 and with R—2 as a grid leak. R—3 is the plate load resistor for the tube T—1.

The output of T—1 is impressed, thru the isolating resistor R—4, on the "consonator" circuit consisting of R—5, L—1 and C—3.

The "consonator" circuit operation has been described in connection with the circuits of Fig. 1 and Fig. 4.

The output of the "consonator" circuit is mixed electronically, by means of T—2 with the amplified output of the microphone M—2. The mixed signals are filtered of their extreme high frequency components by F—2 and transmitted so as to utilize the advantages of the invention.

Under certain conditions, it has been found advantageous to incorporate, after the oscillator O—2 and before T—2, a volume control tube, operated thru a D. C. phase inverter tube, by the rectified D. C. voltage which is impressed on the number 2 grid of T—7. This control tube would lower, quantitatively, the amount of signal applied to grid number 1 of tube T—2 when the brilliancy of the sound source at microphone M—1, diminishes. This has not been shown in the drawing since its insertion in the circuit is believed clear without illustration.

Fig. 6 shows another alternate for the "complexity" generator and the "consonator" circuit. The input E—1 consists of an audio frequency signal preferably subjected to a high pass filter similar in characteristics to F—1 shown in Fig. 1 and already described. The signals E—1 are transformer coupled to a grid leak detector T—4. The vacuum tube T—4, advisedly, should be of the remote cut-off type operated with a low plate load resistor. The time constant (R—12) × (C—10), when selected as of the order of 10 milliseconds, will yield good results.

The condenser C—2 serves to couple the output of T—4 to the "consonator" circuit, which in this case, is the entire remaining portion of the circuit. The action of the "consonator" circuit of Fig. 6 is similar to the action of previously described "C" circuits, with the exception of that shown by Fig. 3. In Figs. 1 and 2, the selection and phase shift upon which the operation of the "consonator" circuit is based, is performed by the tuned circuit L—1, C—3 and the variation in the susceptance component of the input admittance of the tube T—2. The magnitude of this susceptance is a function of the grid to plate capacity and the transconductance of the tube with respect to the grid input which is connected across the tuned circuit. A variation in the transconductance of the tube T—3, such as that caused by a change in bias on one grid, will effect the input admittance. The resulting change in the susceptance term however is accompanied by a change in the conductance term of the admittance. This latter effect produces a form of amplitude modulation of the signals impressed across the tuned circuit. On the other hand, the circuit of the vacuum tube T—3 in Fig. 6 has produced results without much amplitude modulation. It is not deemed necessary to go into detail concerning this circuit since it will immediately be recognized as one commonly used as the reactance tube in automatic frequency control circuits in current radio receivers.

If the feedback circuit R—16, C—14, C—15 and R—17 is arranged, in conjunction with a high transconductance vacuum tube T—3, to feed back a large portion of the voltage appearing across the tuned circuit, then the scanning process, as accomplished by causing the transconductance of the tube to vary at a low audio rate, need not be used. For the large feedback conditions the tube T—3 with its associated circuit acts to shift the phase of the incoming signal (that applied across the tuned circuit) above and below the resonance frequency of the tuned circuit. This phase-shift, due to the amplitude compensation of the feedback, is accomplished without producing an annoying resonance peak.

The output E—2 may be mixed with an auxiliary low frequency channel in the manner of Fig. 1 and the combined signals thereafter transmitted to the utilization apparatus.

A variation of the circuit of Fig. 6 which has produced exceedingly good results is shown in Fig. 7. Here the high frequency together with attenuated low frequency portions of a signal are applied, thru the transformer TR—1, to the "complexity" generator tube T—4. The "complexity" generator is shown in Fig. 7 as a grid leak detector but it can be replaced by any of the other forms mentioned hereto or hereafter. A slightly unbalanced demodulator circuit such as that of Fig. 2 has given excellent results. The tube or tubes used for this purpose, namely, the "complexity" generator should preferably be of the variable mu or remote cut-off type.

The output of the "complexity" generator is applied thru the coupling condenser C—2 to the "consonator" tube T—10. A tuned circuit L—1 and C—3 is used as the plate load for T—10. A feedback circuit consisting of the network R—19, C—17, C—16 and R—18 is used to apply a phase shift and also degenerate the voltage appearing across the plate load of T—10. This feedback circuit is so arranged as to introduce a voltage on the grid of T—10 which is leading the plate signal of T—10 by approximately 90 degrees, thereby causing the tube T—10 to act as a variable inductance in shunt with the inductance L—1. The value of this shunt inductance is dependent and is a function of the amplification of tube T—10. The low frequency modulations in the output of tube T—4 therefore cause a change in the transconductance of tube T—10, thereby bringing about a transitional low frequency shifting or tuning of the plate circuit impedance of tube T—10. The condenser C—16 is of such a value as to prevent the amplified low frequency modulations in the plate circuit of T—10 from being degenerated or fed back to its grid. The values of the feedback circuit can be so selected that the resonance rise across the tuned plate circuit of T—10 is constant over approximately a 1,500-cycle band; as an example, from 2,500 cycles to 4,000 cycles. This is done by making the phase shift from the plate of T—10 through the feedback circuit to the grid of T—10 less than 90 degrees, thus bringing about an amplitude degeneration which tends to make the voltage across the plate circuit of T—10 more independent of frequency. This circuit of Fig. 7 has produced transitional phase modulation which, within limits, is very nearly free from amplitude modulation. In the adjustment of this circuit, it may be advisable to apply a steady frequency of approximately 3,000 cycles to the grid of T—10 and check the adjustments of the circuit by varying the transconductance independently, by, for example, varying the "C" bias applied to the grid of T—10.

If the input circuit from T—4 to T—10 is of low impedance, it may be necessary to insert an isolating resistance or circuit between the output of tube T—4 and the input of T—10; or the same purpose may be accomplished by connecting the output of the condenser C—2 to a tap on R—18 instead of connecting it to the high side of R—18. T—10 is a high mu triode or pentode or, if the resulting phase shift is compensated for, a multi-tube amplifier may be used.

The effect of this arrangement (T—10 in Fig. 7) is to produce a fairly rapid phase shift vs. frequency characteristic in the complexity pattern generated by T—4.

Good results have been obtained by setting the resonance point of the tuned circuit slightly below the highest frequency that the subsequent apparatus is capable of transmitting. The output of the "consonator" circuit is taken across the tuned circuit as shown in Fig. 7. The output E—2 can be mixed with a low and medium frequency channel before transmission to the recording apparatus, radio transmitter or whatever medium is chosen as a medium of ultimate transmission to an electro-acoustic converter and to the ear.

In the circuits illustrated and described above, only a few of the possible ramifications have been specifically elucidated. As an example, the "complexity" generator has been shown as a grid leak detector, a push-pull tube detector, biased detector, push-pull crystal rectifiers and a medium frequency audio signal frequency modulated and subjected to the non-linearity of a grid leak detector. Multi-stage combinations of these various individual circuits have also been used with excellent results. The basic function of the "complexity" generator is to produce a complex, multi-frequency audio pattern; the prime factor of the frequencies of this audio pattern being a low number. The prime factor need definitely not be a fixed factor but may increase or decrease as a function of time and as a function of amplitude of the input signal. When the high frequency portion of a signal is used to supply the basis for the generation of the complex pattern by a non-linear detector, in general, changes in the prime factor of the complex pattern component frequencies will be carried for by the changing complexity of the input signals due to the transients and changes in the magnitude and number of frequencies in the basic high frequency signal.

To the skilled technician, there are numerous methods and processes for generating the complex audio pattern desired. The process may even be carried on at a radio frequency band and heterodyned into the useful audio range. These various changes, modifications, and diversifications in the form and details of the circuits illustrated and their operation, it will be understood, may readily be made by those skilled in the art without departing from the spirit of the invention.

Several specific arrangements for the "consonator" circuit have been described hereto. The function of the "consonator" circuit has also been explained. To those skilled in the art, ways and means of accomplishing the same effect and result, now that the basic principles have been disclosed, will be evident. It will be ascertained from the above that the invention is not limited to the particular arrangements of parts and circuits disclosed and methods described herein for illustration, but that the manifest novel concept and basic inventive ideas are susceptible to numerous variations and modifications coming within the broad scope and spirit of the invention as defined in the appended claims.

I claim as my invention:

1. In an electrical circuit for the production of subjective auditory effects and including a non-linear distorting circuit element with impressed audio frequency modulations, means for selecting at least a part of the output of said non-linear circuit element, means for transitionally modifying said selected part as a function of time so as to produce a plurality of audio frequency sidebands, and means for collecting and transmitting at least a portion of the said transitionally modified, selected, part.

2. In an electrical circuit for the production of subjective auditory effects and including a source of audio frequency modulations, means for selecting at least a part of the audio frequency modulations, means for transitionally shifting the phases of the selected part as a function of time so as to produce a plurality of audio frequency sidebands, and means for collecting and transmitting at least a portion of the thus transitionally phase shifted, selected, part.

3. In an electrical circuit for the production of subjective auditory effects and including a source of audio signal modulations, a method which comprises the steps of: selecting at least a portion of the audio signal modulations, modifying said selected portion by the action of a non-linear circuit element so as to produce a signal having a plurality of audio intermodulation frequencies, selecting from the output of

said non-linear circuit element substantially those frequency components which fall within predetermined frequency limits, varying the said predetermined frequency limits as a function of time, and selecting and transmitting at least a portion of the thus transitionally selected frequencies.

4. A method of producing, in an electrical circuit, a signal for subjective auditory effects and comprising the steps of: selecting at least a portion of an audio signal modulation, modifying said selected portion by the action of a non-linear circuit element so as to produce a plurality of audio intermodulation frequencies, selecting from the output of said non-linear circuit element substantially those frequency components which fall within predetermined frequency limits, varying the phases of said selected frequency components as a function of time, and choosing and transmitting at least a part of the signals resulting from said transitional phase variation.

5. A method of generating, in an electrical circuit, a signal for the production of auditory subjective effects and comprising the steps of: selecting at least a part of an audio signal modulation, modifying the said selected part by the action of a non-linear circuit element so as to produce a plurality of audio intermodulation frequencies, selecting from the output of said non-linear circuit element substantially those frequency components which fall within predetermined frequency limits, varying the said predetermined frequency limits as a function of time, collecting at least a portion of the frequencies resulting from such transitional frequency selection, shifting the phases of the said collected frequencies as a function of time and transmitting at least a part of the frequencies resulting from the transitional phase-shifting of said collected frequencies.

6. A method of generating, in an electrical circuit, a signal for the production of subjective auditory effects and comprising the steps of: selecting at least a part of an audio signal modulation, modifying the selected part by the action of a non-linear circuit element so as to produce a plurality of intermodulation frequencies, selecting at least a part of the component frequencies from the output of said non-linear circuit element, shifting the phases of the component frequencies selected from the output of the said non-linear circuit element as a function of time, choosing those frequencies resulting from said transitional phase-shifting as fall substantially within predetermined frequency limits, varying the predetermined frequency limits as a function of time, and collecting and transmitting at least a part of the frequencies thus selected.

7. A method of generating, in an electrical circuit, a signal for the production of subjective auditory effects and comprising the steps of: selecting a portion of an audio signal modulation falling substantially within predetermined frequency limits, varying the predetermined frequency limits within which said selection is accomplished as a function of time so as to produce a plurality of audio frequency sidebands, and collecting and transmitting at least a portion of the frequencies resulting from the said transitional selection.

8. A method of generating, in an electrical circuit, a signal for the production of subjective auditory effects and comprising the steps of: selecting at least a portion of an audio signal modulation, varying the phases of the component frequencies of said selected portion as a function of time so as to produce a plurality of audio frequency sidebands, and collecting and transmitting at least a part of the signal resulting from the said transitional phase variation.

9. A method of increasing the apparent frequency range of a transmitted audio signal comprising the steps of: selecting at least a part of the audio frequency modulations, dividing this selected part into two channels, modifying the first of said divided channels by the action of a distorting device, subjecting the second of said divided channels to a phase-shifting circuit, modifying the phase-shifted signals by the action of a second distorting device, modulating the amplitude of said modified, phase-shifted signals as a function of time, collecting and adding at least a part of the component frequencies of the said first divided, modified channel, to at least a part of the phase-shifted, modified and modulated second channel and transmitting at least a portion of the summated resulting frequencies.

10. A method of increasing the apparent frequency range of a transmitted audio signal comprising the steps of: selecting at least a part of the audio frequency modulations, dividing this selected part into two channels, modifying the first of said two channels by the action of a phase-shifting circuit, collecting the frequencies of said phase-shifted first channel, modulating the amplitudes of the said collected frequencies as a function of time, collecting at least a part of the second of said two channels, adding the collected part of the second of said two channels to the frequencies resulting from the modulation of the collected phase-shifted components of the first channel and transmitting at least a portion of the summated, resulting frequencies.

11. In a system of the class described for the pseudo-extension of frequency bands and containing a non-linear device upon whose input are impressed audio frequency modulations, means for generating an audio signal, means for varying the amplitude of said audio signal, and means for introducing said variable audio signal to the input of the non-linear device thereby causing inter-modulation frequencies between said variable audio signal, and the audio frequency modulations coincidentally impressed on said input of the non-linear device, to be produced as part of the output signal of said non-linear device.

12. In a system of the class described for the pseudo-extension of frequency bands and containing a non-linear device upon whose input are impressed audio frequency modulations, means for creating a complex spectrum of audio frequencies, means for varying the number and magnitude of the frequency components of said complex audio spectrum, means for introducing said variable complex spectrum to the input of the non-linear device thereby causing inter-modulation frequencies between said variable complex audio spectrum, and the audio frequency modulations coincidentally impressed on said input of the non-linear device to be produced as part of the output signal of said non-linear device.

13. A method of increasing the auditory perceptivity of a transmitted audio signal according to claim 4 including the step of varying the amplitude of the ultimate, phase varied, transmitted signal in conformity with changes in amplitude of the input audio signal modulations.

14. An electrical circuit for the production of subjective auditory effects and including a source

of audio signal modulations, an input high pass filter, a non-linear circuit element, a tuned resonance circuit, a low frequency auxiliary signal, a variable impedance circuit element and a low pass filter, and including means for impressing the audio signal modulations on the input of the input high pass filter, means for impressing the output of the input high pass filter on the input of the non-linear circuit element, means for impressing the output of the non-linear circuit element on the tuned resonance circuit, means for varying the impedance of the variable impedance circuit element at the frequency of the low frequency auxiliary signal, means for varying the impedance of the tuned resonance circuit by the transitionally varied variable impedance circuit element, means for impressing the signal appearing across the tuned resonance circuit on the input of the low pass filter, means for transmitting the output signal of the low pass filter and means for rendering the transmitted signal audible.

15. In a signal translating circuit for increasing the auditory perceptivity of an audio signal according to claim 1 and including means for selecting and transmitting at least a part of the input, impressed, audio frequency modulations.

16. In a signal translating circuit for increasing the auditory perceptivity of an audio signal according to claim 2 and including means for selecting and transmitting at least a part of the input audio frequency modulations.

17. A method of increasing the auditory perceptivity of a transmitted audio signal according to claim 3 and including the step of selecting and transmitting at least a part of the input audio frequency modulations.

18. A method of increasing the auditory perceptivity of a transmitted audio signal according to claim 4 and including the step of selecting and transmitting at least a part of the input audio frequency modulations.

19. A method of increasing the auditory perceptivity of a transmitted audio signal according to claim 5 and including the step of selecting and transmitting at least a part of the input audio frequency modulations.

20. A method of increasing the apparent frequency range of a transmitted audio signal according to claim 6 and including the step of selecting and transmitting at least a part of the input audio frequency modulations.

21. A signal translating circuit according to claim 14 and including means for selecting and combining at least a part of the input audio signal modulations with the signal applied to the input of the output low pass filter.

LOUIS A. DE ROSA.