# United States Patent

[11] 3,626,077


[72] Inventors **Walter Munch, Jr.; William S. Wagner; Dale M. Huetrecht, all of c/o D. H. Baldwin Co., 1801 Gilbert Ave., Cincinnati, Ohio 45202**



[21] Appl. No. **40,536**
[22] Filed **May 26, 1970**
[45] Patented **Dec. 7, 1971**


[54] **ORGAN TONE MODULATION SYSTEM**
**7 Claims, 6 Drawing Figs.**


[52] U.S. Cl. ..... **84/1.24,** 84/1.01, 84/1.11
[51] **Int. Cl.** ..... **G10h 1/02**
[50] **Field of Search** ..... 84/DIG. 4, 1.24, 1.11, 1.01; 331/57, 45; 328/24; 321/51, 53–54



[56] **References Cited**

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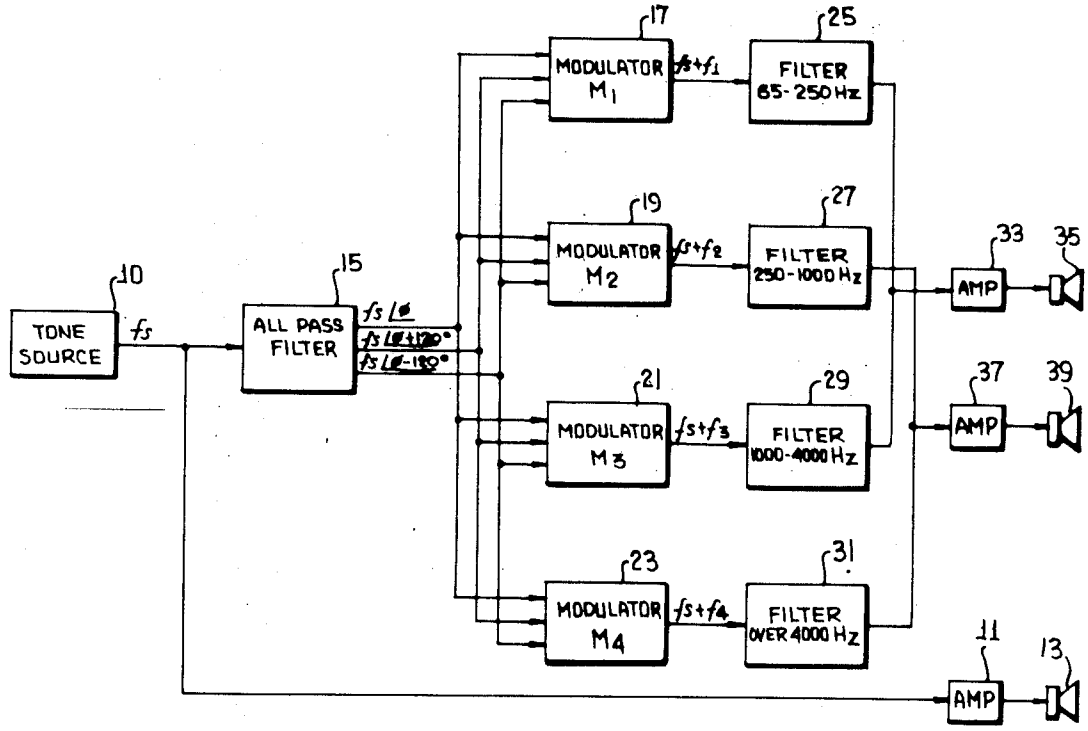
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*Primary Examiner*—D. F. Duggan
*Assistant Examiner*—Ulysses Weldon
*Attorneys*—W. H. Breunig and Hurvitz & Rose



**ABSTRACT:** Organ output, comprising a band of audio frequencies, is divided into four frequency subbands, each of which is shifted in frequency by one of four respective diverse increments. The first and third subbands are acoustically transduced and radiated via a rotary loudspeaker and the second and fourth subbands via a stationary loudspeaker. These radiations are acoustically mixed with the radiated unmodified organ output to produce desired tonal effects. Frequency shifting of the subbands can be inhibited at will, means being provided automatically to maintain the acoustic signal level the same whether or not frequency shifting is in force, for a given setting of the expression pedal. Each of the four subbands can be shifted by either of two respective frequency increments, or inhibited, to permit selective variation of the overall acoustic effect. Frequency shifting for each subband is achieved by dividing the organ tone into three components of equal magnitude and 120° phase difference. To obtain the 120° spaced components, the input tone signal is applied to two-phase shift filters which provide two of the components, the third component being derived by summing the first two components and inverting the resulting signal. Each component is applied to a respective section of four three-section modulators, each section comprising a transistor pair connected to form a DC differential amplifier with one input at AC ground. The emitter currents of each transistor pair are varied at the shift frequency by a three-phase oscillator, each modulator section being controlled by a respective oscillator phase. The outputs of each section of the modulator are combined to cancel the oscillator components provided at each section.

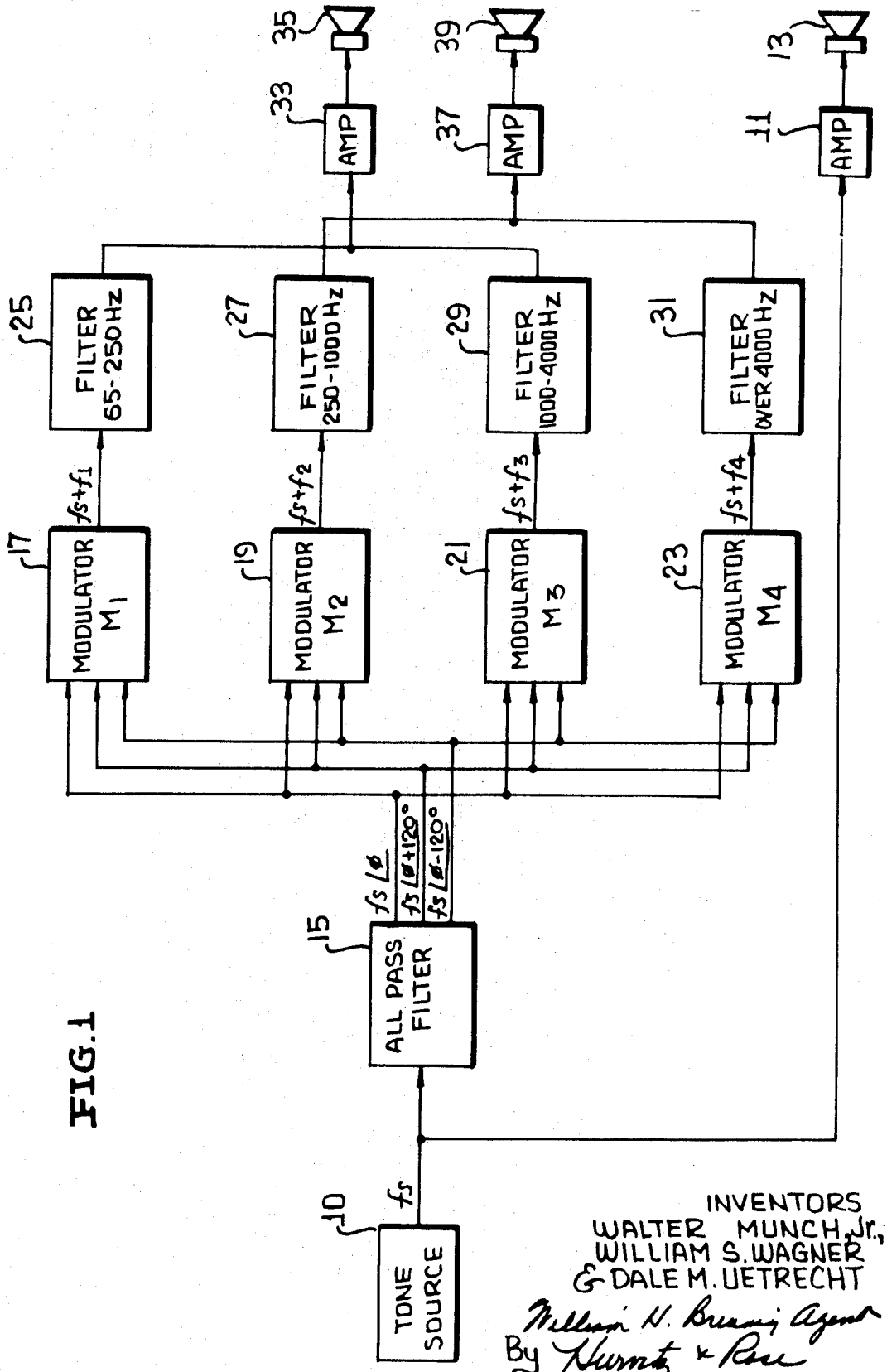
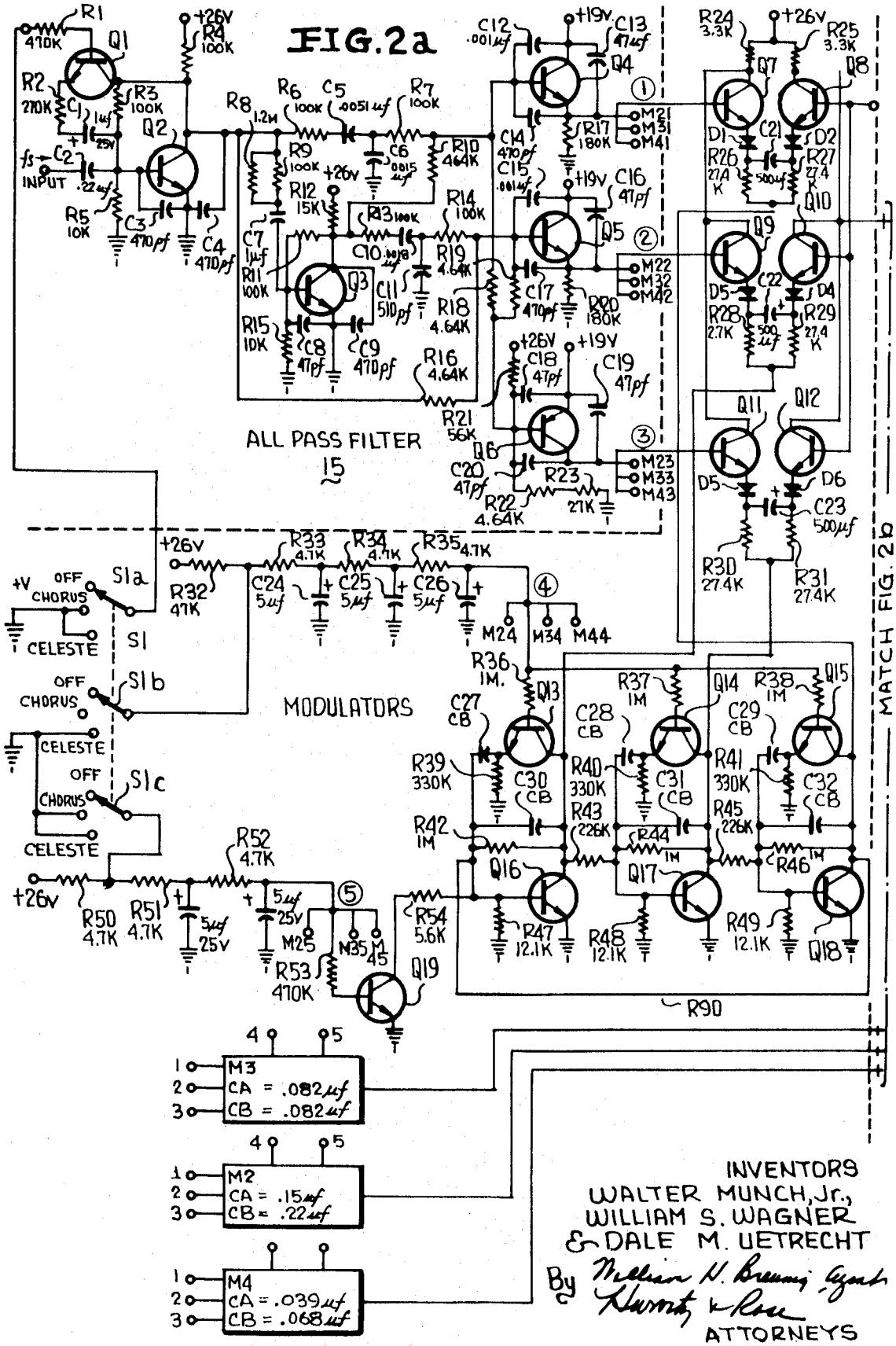



FIG. 1

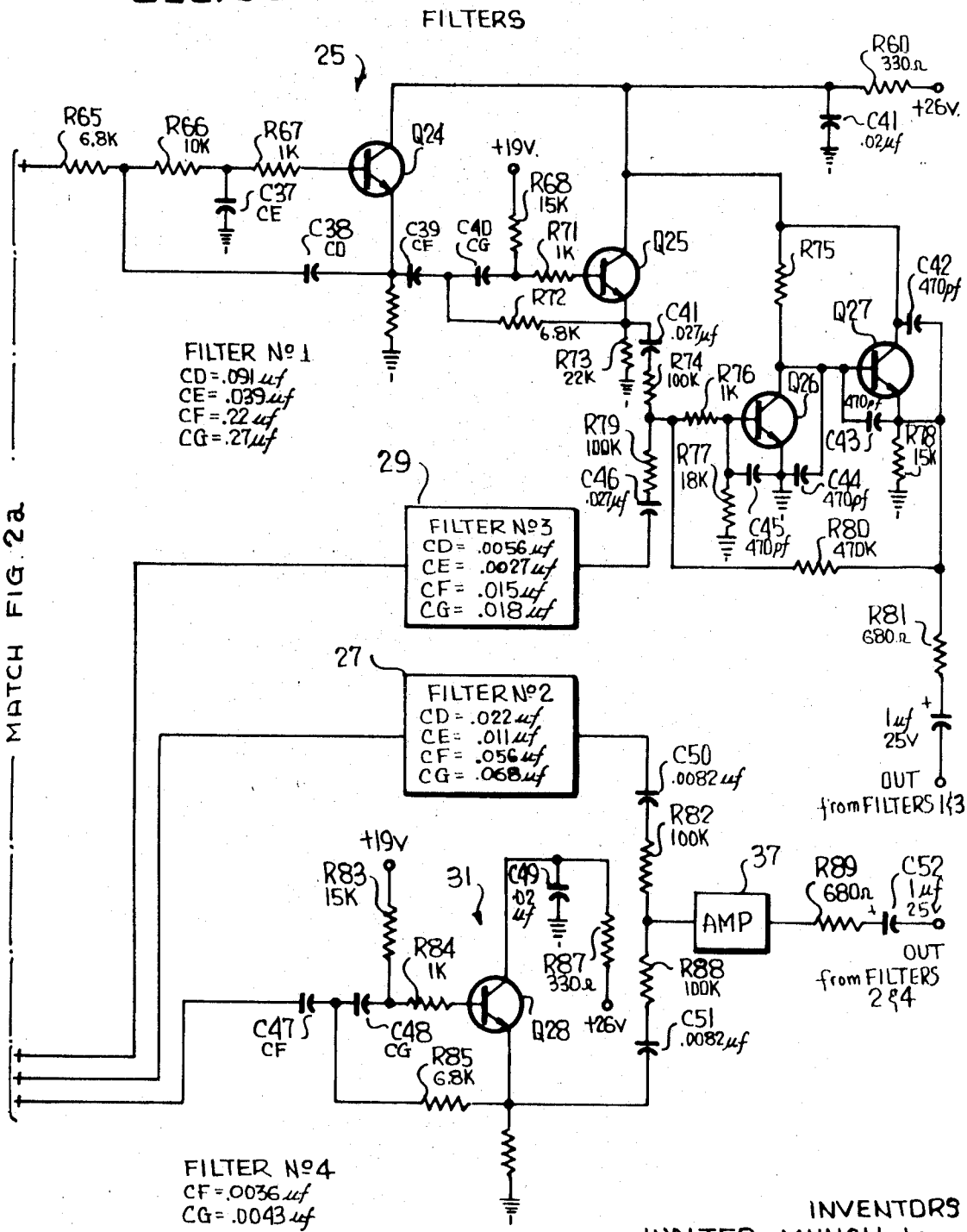
INVENTORS
WALTER MUNCH, Jr.,
WILLIAM S. WAGNER
& DALE M. UETRECHT
By Merriam N. Breen, Agent
Hurwitz & Rose
ATTORNEYS

FIG. 2a




INVENTORS
WALTER MUNCH, Jr.,
WILLIAM S. WAGNER
& DALE M. UETRECHT
By Millen N. Brumm, Agent
Harris & Rose
ATTORNEYS

FIG. 2b




INVENTORS
WALTER MUNCH, Jr.,
WILLIAM S. WAGNER
& DALE M. UETRECHT

By Mellgren W. Breunig, Agents
Hunt & Race
ATTORNEYS

FIG. 3a

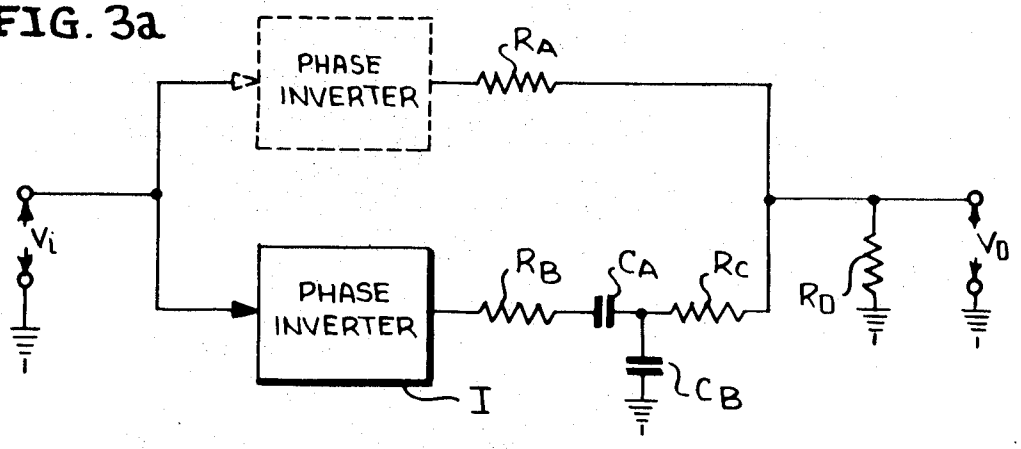



FIG. 3b

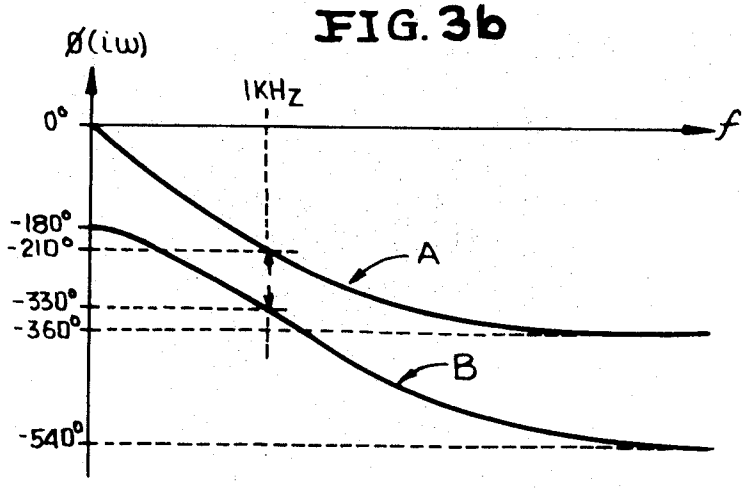
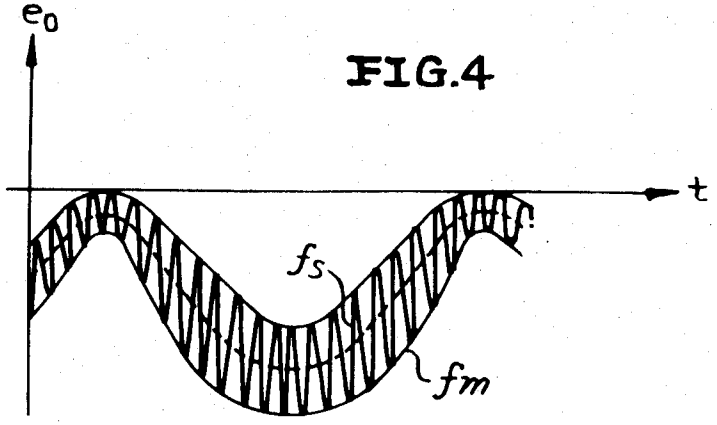



FIG. 4




INVENTORS
WALTER MUNCH, Jr.,
WILLIAM S. WAGNER
& DALE M. UETRECHT

By Williams, Thumm & Rose
ATTORNEYS

# ORGAN TONE MODULATION SYSTEM

## BACKGROUND OF THE INVENTION

The present invention relates generally to systems for achieving an ensemble effect in music by processing a band of audio frequencies, and more particularly by translating the frequency spectrum of musical tones to achieve pleasurable effects. Still more particularly, the present invention is an improvement of the audio modulation system disclosed in U.S. Pat. No. 3,147,333 to Wayne, Jr.

The traditional pipe organ consists of many ranks of pipes which are invariably slightly out of tune. Certain ranks of pipes, referred to as celestes, are purposely detuned by a considerable amount to produce rich ensemble effects in the composite tone, which effects are especially desirable for use in ecclesiastical music. The prior art is replete with examples of attempts to electronically simulate these pleasurable ensemble effects. One such example may be found in the aforementioned Wayne, Jr. patent, which discloses a system in which an organ tone signal is separated into four octavally related frequency subbands, each of which is shifted in frequency by a different amount. The shifted frequency subbands are acoustically radiated, and the resulting sound, when acoustically mixed with the sound of the unmodified organ tone, produces pleasing ensemble effects.

In the performance of certain musical compositions it is often desirable to eliminate the ensemble effects yet still employ the loudspeakers associated with the ensemble system. A problem arises in this regard in a system of the type disclosed in the Wayne, Jr. patent because the system gain decreases considerably when the modulator circuits are turned off. One object of the present invention is to provide means for maintaining a constant volume level in a system such as that disclosed by Wayne, Jr., whether or not the modulator is operative.

Another area in which the Wayne, Jr. system requires improvement concerns the fact that none of the subbands can be shifted by more than one increment. This produces a pleasing acoustic effect but limits the flexibility of the system. It is therefore an object of the present invention to provide means for selectively shifting each frequency subband in the Wayne, Jr. system by a plurality of increments. More particularly, it is an object of the present invention to permit shifting of each frequency subband by at least two individually selected increments.

The basic technique employed in the Wayne, Jr. system to obtain the four frequency subbands is to provide three components of the input tone signal, each being of equal amplitude but separated 120° in phase. Each of these components is applied to each of four tone modulators at which all three components are modulated by four respective subsonic modulation signals. The three components are then recombined in each modulator and applied to a respective band-pass filter to provide the four resulting frequency shifted subbands. The Wayne, Jr. system therefore requires a circuit capable of segmenting a wideband signal into three like signals of equal amplitude but spaced 120° in phase. Likewise the system requires modulators capable of shifting these components by subsonic frequency increments. It is an object of the present invention to provide simple and reliable circuits capable of effectively performing these functions in a manner consistent with the intended operation of the Wayne, Jr. system. In addition, it is an object of the present invention to provide circuits capable of performing these functions and yet which are able to pass the input tone signal through to the loudspeakers with no change in gain when frequency shifting is not desired.

It is still another object of the present invention to provide improvements in the circuitry of the Wayne, Jr. system in order to provide new and desirable functions.

## SUMMARY OF THE INVENTION

In accordance with one aspect of the present invention there is provided a system for achieving ensemble effects in music by separating a frequency spectrum into octaval subbands, each of which is shifted by a separate discrete frequency increment, and recombining the subbands both electrically and acoustically such that two nonadjacent subbands are radiated from a stationary loudspeaker and two nonadjacent subbands are radiated from a rotary loudspeaker. The incoming tone signal is supplied to two phase-shift circuits to provide two signal components of equal amplitude but shifted in phase by 120°. The two components are then summed and inverted to provide a third component with the same amplitude but spaced 120° from the first two. The three components are applied to each of four three-phase modulator circuits, the modulation frequency being different but subsonic. Each modulation circuit includes a three-phase subsonic oscillator and three DC differential amplifiers, one of each for each of the three input signal components. The current through each differential amplifier is modulated by a respective subsonic oscillator phase. The output signals of all three differential amplifiers are combined to cancel out the subsonic modulation frequency components and produce a resultant signal which is an amplified version of the input tone signal applied to the system, frequency shifted by the modulation frequency.

The output signal from each modulator circuit is applied to a respective band-pass filter which provides the required octaval separation to produce the desired frequency subbands. The first and third octaval bands are combined and applied to the rotary speaker whereas the second and fourth octaval bands are combined and applied to the stationary speaker.

When frequency-shifting is not desired the oscillators providing the 120° phase shifted subsonic modulation signals are rendered nonoscillatory; however, one DC differential amplifier in each of the four modulators is biased on while the other two are biased off to thereby permit one phase component of the input signal to be passed through each modulator. The increase in gain occurring as a consequence of turning off two sections in each modulator is compensated for by decreasing the gain of a preamplifier for the input tone signal before it is separated into the three phase components.

The three subsonic oscillators associated with each modulator are primarily amplifiers with respective RC phase shift circuits connected thereto to produce oscillation. By selective actuation of a switch the capacitance of the phase shift circuitry in each group amplifiers is varied by the same amount to thereby alter the subsonic modulation frequency provided by these circuits.

## BRIEF DESCRIPTION OF THE DRAWINGS

The above and still further objects, features and advantages of the present invention will become apparent upon consideration of the following detailed description of specific embodiments thereof, especially when taken in conjunction with the accompanying drawings, wherein:

FIG. **1** is a block diagram of the system according to the present invention;

FIGS. **2***a* and **2***b* combined represent a schematic circuit diagram of a preferred embodiment of the present invention;

FIG. **3***a* is a schematic circuit diagram of a single section of the all-pass filter network of FIGS. **2***a*, **2***b*;

FIG. **3**is a plot of the phase versus frequency characteristic of the circuit of FIG. **3***a*; and

FIG. **4** is a plot versus time of the output signal from a single phase of the three phase modulator of the present invention.

## DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring now more particularly to FIG. **1** of the accompanying drawing there is illustrated a system of the type disclosed in the aforementioned Wayne, Jr. patent wherein a wide band audio signal having a frequency fs is derived from a tone source **10** representing, for example, an electronic organ. Signal $f_s$ is applied to an amplifier **11** and in turn to a loudspeaker **13** in a conventional manner, and in addition is applied to an all-pass filter network **15**. The latter provides, on

three separate output leads, three respective components of the original $f_s$, the components having equal amplitudes but mutual phase differences of 120°.

The three components of signal $f_s$ are identically treated, each being applied to each of four modulator circuits **17**, **19**, **21** and **23** which are also designated $M_1$, $M_2$, $M_3$ and $M_4$, respectively. These modulators are of the single side band type, each introducing a different frequency shift $f_1$, $f_2$, $f_3$, and $f_4$ respectively, in the original signal $f_s$. The output signals from the modulators are thus designated $f_s+f_1$, $f_s+f_2$, $f_s+f_3$, and $f_s+f_4$, respectively. Each of the modulators includes means for generating a three-phase subsonic modulation signal at frequencies $f_1$, $f_2$, $f_3$, and $f_4$ respectively, one phase for each of the three phases of the input signal received from all-pass filter network **15**. The resulting modulator output signals are applied to respective filters **25**, **27**, **29** and **31**. These filters act to separate the four modulator output signals into respective octavely related subbands. For example, filter **25**, which has its 3 db. points at 65 Hz. and 250 Hz. receives the signal $f_s+f_1$ from $M_1$ and represents the lowest octave subband. Consequently, modulation frequency $f_1$ is the lowest of the four modulation frequencies. The pass band of filter **27** is 250 to 1,000 Hz., that of filter **29** is 1,000 to 4,000 Hz., and that of filter **31** is 4,000 Hz. and above.

The output signals from filters **25** and **29** are electrically combined in amplifier **33** and acoustically radiated by loudspeaker **35**. Likewise, the output signals from filters **27** and **31** are electrically combined in amplifier **37** and acoustically radiated via loudspeaker **39**. Loudspeakers **35** and **39**, along with loudspeaker **13**, are well spaced and provide desirable acoustic mixing. Undesired beats between the acoustically radiated signals are avoided by applying adjacent subbands to different loudspeakers. In addition, one of loudspeakers **35** and **39**, for example loudspeaker **35**, is preferably of the rotary type whereas the other is stationary.

The system as thus far described is essentially the Wayne, Jr. system disclosed in the aforementioned U.S. Pat. No. 3,147,333. The system of FIG. **1** also serves as a block diagrammatic representation of the system of the present invention; however, the improvements afforded by the present invention are clearly specified in the following description of FIGS. **2***a* and **2***b*.

The following brief mathematical analysis summarizes the approach employed to achieve the desired shifts in the frequency of $f_s$. Three signals at frequency $f_s$, of equal amplitude and separated in phase by 120°, are represented by the following expressions:

$$\sin \omega_s t \quad (1)$$

$$\sin (\omega_s t+120°) \quad (2)$$

$$\sin (\omega_s t-120°) \quad (3)$$

where $\omega_s=2\pi f_s$, and $t$ represents time. If these signals are amplitude modulated by three signals having a frequency $f_m$, the modulation signals also being separated by 120° in phase, signals represented by the following expressions, respectively, result:

$$\sin \omega_s t[1+m \sin \omega_m t] \quad (4)$$

$$\sin (\omega_s t+120°)[1+m \sin (\omega_m t-120°)] \quad (5)$$

$$\sin (\omega_s t-120°)[1+m \sin (\omega_m t+120°)] \quad (6)$$

where $\omega_m=2\pi f_m$, and m is a modulation constant associated with $f_m$.

Expressions (4), (5), and (6) may be expanded by trigometric identities to provide the following expressions, respectively:

$$\sin \omega_s t+m/2 \cos (\omega_s-m/2 \cos (\omega_s+\omega_m) t \quad (7)$$

$$\sin [\omega_s t+120°]+m/2 \cos[(\omega_s-\omega_m)t+240°] -m/2 \cos(\omega_s+\omega_m) t \quad (8)$$

$$\sin [\omega_s t-120°]+m/2 \cos [(\omega_s-\omega_m)t-240°] -m/2 \cos (\omega_s+\omega_m)t \quad (9)$$

If expressions (7), (8) and (9) are summed, the first terms of each cancel one another, the second terms of each cancel one another, leaving as the sum of the three expressions the following: $-3m/2 \cos (\omega_s+\omega_m)t$ (10)

Expression (10) represents a signal shifted from frequency $f_s$ by an amount $f_m$.

From the above analysis it is clear that the following operations can be performed electrically in order to achieve the desired frequency shift:

a. the incoming signal $f_s$, must be segmented into a three phase signal; that is, into three components of equal amplitude displaced from one another by 120°;

b. an oscillator must be employed having a frequency $f_m$ which also provides a three phase output in which each component has the same amplitude and is displaced from the others by 120°; $f_m$ is subsonic because only very small percentage shifts in the tone frequency can be musically tolerated;

c. the three components of $f_s$ must then be multiplied by appropriate ones of the $f_m$ components; and

d. the three resulting signals must be summed to achieve the desired frequency shift.

Above steps (b), (c) and (d) must be performed at each of modulators **17**, **19**, **21** and **23** to obtain the four different frequency shifted subbands.

The description which follows illustrates in detail how each of the above steps (a) through (d) are performed in the present system.

Referring now particularly to FIG. **2***a* of the accompanying drawings, signal $f_s$ is illustrated as applied to the input terminal of all-pass filter network **15** from which it is AC coupled via capacitor **C2** to the base of NPN-amplifier transistor **Q2**. **Q2** is biased to operate in the class A mode, and as part of its bias circuitry has a resistor **R3** connected between its collector and base. Connected across **R3** is the series combination of the collector-emitter circuit of NPN transistor **Q1**, resistor **R2**, and transient suppression capacitor **C1**. The base of **Q1** is coupled via resistor **R1** to the arm **S1***a* of one section of a three-section selector switch **S1**. Switch **S1**, as described in greater detail below, is capable of switching the system into three operational modes, namely chorus, celeste, and off. In the chorus and celeste modes signal $f_s$ is shifted by different modulation frequencies; in the off mode signal $f_s$ passes through the system without frequency shift. In the chorus and celeste position of switch **S1** the base of **Q1** is grounded and thereby biased off, leaving only **R3** operatively connected across the collector and base of **Q2**. When **S1** is in its off position, positive voltage is applied to the base of **Q1** rendering the latter conductive and effectively connecting **R2** in parallel with **R3** to lower the gain of **Q2**. This permits the volume of the system to be maintained at the same level in the off mode as in the celeste and chorus modes, as described below.

The amplified version of signal $f_s$ appearing at the collector of **Q2** is applied to the filter section of all-pass filter network **15**. The purpose of all-pass filter network **15** is to separate signal $f_s$ into three separate signals of equal magnitude and 120° out of phase with one another. In addition the filter must have a flat amplitude versus frequency response characteristic over the entire audio band of signal $f_s$. The filter includes a second order filter section comprising resistor **R6**, capacitor **C5** and resistor **R7** connected in series and a capacitor **C6** connected from ground to the junction between capacitor **C5** and resistor **R7**. Connected in parallel across the series combination of **R6**, **C5** and **R7** is a pair of parallel connected resistors **R8** and **R9**, connected in series with capacitor **C7**, a phase-inverting NPN-transistor **Q3**, and a resistor **R10**. Phase-inverter **Q3** is also in series with a filter section comprising series connected resistor **R13**, capacitor **C10**, and resistor **R14**, there being a further capacitor **C11** connected between ground and the junction between **C10** and **R14**. A resistor **R16** is connected between the collector of **Q2** and the end of **R14** remote from **C11**. The collector of **Q3** is returned to +26 volts DC via resistor **R12** and to ground via voltage divider resistors **R11** and **R15**, whose junction is connected to the base of **Q3**. The base and collector of **Q3** are coupled to ground via respective capacitors **C8** and **C9**. The collector of **Q3** is also connected to the end of **R7** remote from **C6** via resistor **R10**. The same end of **R7** is connected to the base of a further transistor **Q4**, and the end of **R16** remote from the collector of **Q2** is connected to the base of transistor **Q5**.

Signals appearing at the bases of Q4 and Q5 are of equal amplitude and are separated in phase by 120°. The manner in which this phase separation is accomplished may best be appreciated by reference to FIG. 3*a* of the accompanying drawings wherein a single section of the above described filter is illustrated. Considering only the solid lines for a moment, there is illustrated between the input and output terminals of the filter section a first circuit branch including a phase inverter I, corresponding to Q3 of FIG. 2*a*, connected in series with resistor $R_B$, capacitor $C_A$, and resistor $R_C$, there being a capacitor $C_B$ connected between ground and the junction of capacitor $C_A$ and resistor $R_C$. Across this series circuit there is connected a resistor $R_A$, and the entire parallel combination is returned to ground via load resistor $R_0$ connected to the output terminal. $R_A$ corresponds to resistor R16 in FIG. 2*a*; similar correspondence between the elements of FIGS. 2*a* and 3*a* is had between $R_B$ and R13, $R_C$ and R14, $C_A$ and C10, and $C_B$ and C11. The gain G of the circuit of FIG. 3*a* in terms of the LaPlace operator *s* is given by the following expression:

$$G(s)=\frac{V_0}{V_i}=H\frac{s^2-as+b}{s^2+as+b} \qquad (11)$$

where H, *a*, and *b* are constants whose values are determined by the components $R_A$, $R_B$, $R_C$, $C_A$, $C_B$ and $R_0$, with the value of H additionally depending upon the amplification factor of phase invertor I. Expression (11) represents a second order system having a phase plot as a function of frequency corresponding to curve A illustrated in FIG. 3*b*. Note that the phase for this circuit is zero at zero frequency and decays asymptotically toward 360°.

Now consider the dotted line version of FIG. 3*a* wherein phase inverter I is inserted in series with resistor $R_A$ instead of with the resistor-capacitor network. Such a configuration corresponds to that in FIG. 2*a* wherein $R_A$ now corresponds to R10, the inverter I still corresponds to Q3, and correspondence is additionally had between $R_B$ and R6, $C_A$ and C5, $R_C$ and R7, and $C_B$ and C6. The expression for the gain of this circuit, as a function of *s*, is identical to expression (11) except that a minus sign appears in front of that expression. This negative expression results in a phase plot, corresponding to curve B of FIG. 3*b*, which begins at −180° for zero frequency and decays asymptotically toward −540°. By proper selection of the resistors and capacitors in the solid line embodiment of FIG. 3*a*, curve A in FIG. 3*b* may be adjusted to have a phase shift of −210° at 1 kHz., as illustrated. Likewise the angle of the transfer function for the dotted line embodiment of FIG. 3*a* can be adjusted to provide a phase angle of −330° at 1 kHz. The output signals of the two alternative filter embodiments of FIG. 3*a* can thus be spaced by a 120° phase angle at one kHz. and are maintained within 10° of being 120° out of phase for about 2½ octaves on either side of 1 kHz.; this is true in spite of the fact that curves A and B are 180° out of phase at zero and 180° out of phase at infinity on the frequency scale.

The two phase shift networks described above are illustrated in FIG. 2*a* as sharing a common phase inverter, namely that comprising transistor Q3. The two circuits apply their respective 120° phase-displaced signals to the bases of transistors Q4 and Q5 as described above. Q4 and Q5 are NPN transistors connected in emitter-follower configuration, and provide respective 120° spaced signals at their emitter electrodes without further phase alteration. In order to obtain three signal phases, the two signals applied to Q4 and Q5 are combined via resistors R18 and R19, respectively, at the base of PNP transistor Q6. This serves to effectively sum the two individual components at the base of Q6. The output signal from Q6 is taken from its collector and is therefore an inverted version of the sum of the signals applied to the base of Q6. That is, the two 120° spaced components have been summed and inverted, a procedure which results in a third component spaced 120° from the first two. The gains of amplifiers Q4, Q5, and Q6 are selected to assure that the three 120° spaced components supplied thereby are of equal amplitude.

The all-pass filter network, when provided with the exemplary component values indicated on the drawing, provides the three output signals, spaced in phase as desired, and having equal magnitudes over approximately a 5 octave range (175 Hz. to 5,400 Hz.) with a center frequency of approximately 1,000 Hz.

In accordance with the principles described in relation to FIG. 1 hereinabove, each of the output signals from Q4, Q5, and Q6 is conducted along four individual lines by which it is applied to each of the four modulator circuits $M_1$, $M_2$, $M_3$, and $M_4$ respectively. Since all four modulators are identical except for the modulation frequency employed in each, only one such circuit, namely $M_1$ is illustrated in detail, the others being designated by blocks in FIG. 2*a*. Each of the modulators is illustrated as having five input terminals numbered 1 through 5 respectively, these terminals being employed as the second number in a subscript to designate signal destination. More particularly, the output signal from Q4 is directly applied to terminal 1 of modulator $M_1$ as well as to destinations indicated as $M_{21}$, $M_{31}$, and $M_{41}$. These latter designations mean that this signal is applied to modulator $M_2$, terminal 1; to modulator $M_3$, terminal 1; and to modulator $M_4$, terminal 1. Similar references appear at the output terminals of Q5 and Q6 to indicate the modulator and the terminal number destinations for these signals. Like references appear for signals applied to terminals 4 and 5 of the various modulators, which signals are described in detail hereinbelow.

Referring now in detail to modulator $M_1$, each phase of the three-phase input signal is applied to a respective DC differential amplifier having its second input terminal at AC ground, via its voltage source (not shown). The first phase of the input signal, for example, is connected to the base of NPN-transistor Q7 which has its collector returned to +26 volts DC via resistor R24 and its emitter connected to the anode of diode D1. A similar NPN-transistor Q8 has its base tied to +18 volts DC, its collector tied to +26 volts DC via resistor R25, and its emitter connected to the anode of diode D2. The cathodes of D1 and D2 are connected to opposite ends of capacitor C21 and also via respective resistors R26 and R27 to one section of a three-phase subsonic oscillator to be described in detail below.

Phase two of the input signal is connected to a similar DC differential amplifier having transistors Q9, Q10, diodes D3, D4, capacitor C22, and resistors R28 and R29 which correspond respectively to transistors Q7, Q8, diodes D1, D2, capacitor C21, and resistors R26 and R27. Likewise phase three of the input signal is applied to a differential amplifier having corresponding components in transistors Q11, Q12, diodes D5, D6, capacitor C23, and resistors R30 and R31. The collectors of Q9 and Q11 share collector resistor R24 with Q7; the collectors of Q10 and Q12 share collector resistor R25 with Q8. Each phase of a three-phase subsonic oscillator varies emitter current flow in a respective differential amplifier, thereby modulating the signals applied to those amplifiers. The three 120°-spaced input signal components are thus seen to be modulated by three respective subsonic modulation signals having the same frequency but also spaced 120° in phase. The three output signals from the differential amplifiers are then summed by tying the collectors of transistors Q8, Q10, and Q12 to a common junction, the summing of these signals resulting in a signal described by expression (10) above.

The three-phase subsonic oscillator comprises three respective phase shift oscillators. More particularly, one section of the subsonic oscillator includes NPN-transistor Q16 having its emitter connected directly to ground and its collector connected to the junction between R28 and R29 in the second differential amplifier section. The base of Q16 is returned to ground via resistor R47 and the parallel combination of R42 and C30 is connected between its base and collector. In addition, there is a series path comprising the collector-emitter circuit of NPN-transistor circuit Q13 and capacitor C27 connected across capacitor C30, and a resistor R39 connected to

ground from the junction between capacitor C27 and transistor Q13. The base of Q13 is connected to +26 volts DC via R36 and a transient suppression filter comprising three sections of series resistors R33, R34, and R35, shunt capacitors C24, C25, and C26, and current limiting resistor R32. The +26 volts DC may be selectively shorted to ground, away from the base of Q13, by means of arm S1*b* of switch S1 whenever switch S1 is in the celeste position. When such is the case, Q13 is nonconductive and C27 is not effective in the oscillator circuit. On the other hand, when S1 is in the chorus position +26 volts is applied to the base of Q13 which is then rendered conductive to effectively place C27 in parallel with C30. By this mechanism the frequency of the phase shift oscillator associated with Q16 can be selectively altered.

The collector of Q16 is also connected via resistor R43 to the base of Q17 which forms a further section of the three-phase subsonic oscillator. Elements Q14, R37, C28, R40, C31, R44, and R48 correspond identically in circuit configuration and component values to elements Q13, R36, C27, R39, C30, R42, and R47. The collector of Q17 is connected to the base of transistor Q18 via resistor R45, Q18 comprising the active element of the third subsonic oscillator section and having components Q15, R38, C29, R41, C32, R46, and R49 which correspond to respective components comprising the circuit associated with Q17. The collector of Q18 is coupled to the base of Q16 via resistor R90. All three of transistors Q13, Q14 and Q15 are turned on and/or off simultaneously by the action of arm S1*b* of switch S1.

The collector of Q18 is coupled to the junction between resistor R26 and R27; the collector of Q17 is coupled to the junction between resistors R30 and R31.

The three subsonic oscillator sections oscillate when the phase shift across each transistor is 120°, because then the total circuit loop phase shift is 360°. The frequency of oscillation of each is determined by respective resistors R42, R44, R46 in parallel with respective capacitors C30, C31, and C32. In addition capacitors C27, C28, and C29, when made effective in the circuit via Q13, Q14, and Q15, play a part in determining the oscillation frequency of these oscillators. The component values in all three oscillators are identical to assure that each operates at the same subsonic frequency. The approximate frequencies for modulator $M_1$, $M_2$, $M_3$, and $M_4$ in the chorus mode are 1, 2, 4, and 8 Hz. respectively; in the celeste mode the approximate oscillation frequencies are 2, 4, 8 and 16 Hz. respectively. The output voltages of the collectors of each of the Q16, Q17, and Q18 can be represented by the following three expressions, respectively:

$8+8\sin\omega t$ (12)

$8+8\sin(\omega t+120°)$ (13)

$8+8\sin(\omega t+240°)$ (14)

The base of Q16 is coupled via resistor R54 to the collector of Q19, an NPN transistor whose emitter is grounded. The base of Q19 is connected to +26 volts DC via current limiting resistor R50, a transient suppression network and base resistor R53. This 26 volts may be selectively shunted to ground by section S1*c* of switch S1, the arm of which is connected between current limiting resistor R50 and the transient suppression network. When the mode switch is in the off position, a positive voltage is applied to the base of Q19 which is rendered conductive thereby and effectively switches resistor R54 in parallel with base bias resistor R47 for Q16. This maintains transistor Q16 cut off, rendering its collector more positive. This increased positive voltage is coupled to the base of Q17 which is driven thereby into high conduction to in turn reduce its collector voltage sufficiently to bias Q18 off. Thus, in the system off mode, Q16 and Q18 remain off while Q17 remains on; there is no oscillation in this mode. With Q16 thus held off, the emitter currents in Q9 and Q10 are severely limited. Likewise the fact that Q18 is off severely limits the emitter currents in Q7 and Q8. However, with Q17 biased on there is no such severe limitation on the emitter currents at Q11 and Q12. With transistor pairs Q7, Q8, and Q9, Q10 held in low gain condition, substantially all of the signal applied to all-pass filter section 15 is passed via transistor Q6 and amplifier pair Q11, Q12 to the filter and amplifier circuits which follow the modulator circuits. The action described previously in relation to transistor Q1, whereby R2 is effectively switched in parallel with R3 for amplifier Q2 during the off mode of the system, adjusts the gain in the off mode to compensate for the increased gain condition resulting from the effective inhibition of two of the differential amplifiers. More specifically, the decreased amplification in the circuit of transistor Q2 compensates for the amplification increase resulting from the low gain condition of transistors Q7, Q8 and Q9, Q10. In this manner, the signal volume remains substantially constant at the output of the modulator circuit whether or not frequency shifting is being effected.

Each one of the differential amplifiers modulates a different phase of the three-phase signal applied to the modulator. The gains of the differential amplifiers are inversely proportional to their emitter impedances which in turn are inversely proportional to their emitter currents. The emitter currents are directly controlled by the subsonic oscillations and therefore the output signal from each transistor pair is an amplitude modulated signal, achieved as stated. An example of the output signal waveform from one transistor pair appears in FIG. 4. The high frequency input signal $f_s$ is seen to be amplitude modulated and superimposed on the subsonic modulation frequency $f_m$. The modulation frequency components of the three modulated signals are cancelled out when summed at the junction point of the three collectors of transistors Q8, Q10 and Q12.

The output signals from each of modulators $M_1$, $M_2$, $M_3$ and $M_4$ are applied to respective filters 25, 27, 29, and 31, as illustrated in FIG. 2*b*. Filters 25, 27, and 29 are substantially identical with the exception of their component values (and hence their pass bands) and therefore only filter 25 is described in detail. Filters 27 and 29 are illustrated in block form with their appropriate component values specified. Filter 31 is shown in detail because it differs somewhat from the other three. Filter 25 comprises an active low pass filter cascaded with an active high pass filter, the overall result providing an active band-pass filter with a passband gain of one and a controllable selectivity. The attenuation of the filter in the stop bands is 12 db. per octave. The passbands are designated in FIG. 1. The low pass filter in filter 25 comprises transistor Q24 and associated resistor and capacitor elements connected conventionally to provide an active low pass filter. The output signal from low pass filter Q24 is taken from the emitter electrode of Q24 and applied to the high pass filter section comprising transistor Q25 and associated resistive and capacitive elements connected to provide high pass filtering in a conventional manner. The output signal from the high pass filter is taken from the emitter electrode of Q25 and coupled along with the output signal from filter 29 to the base of NPN-transistor Q26. The latter, along with transistor Q27 comprises a wideband amplifier corresponding to amplifier 33 of FIG. 1, the output signal from which is applied to loudspeaker 35.

Filter 31 includes only an active high pass filter in which transistor Q28 is analogous to transistor Q25 of filter 25. The high pass filter output signal is applied along with the signal from filter 27 to amplifier 37 and in turn to speaker 33 as described in reference to FIG. 1.

Active high and low pass filters of the type employed herein are described in detail in U.S. Pat. application, Ser. No. 734,302, filed June 4, 1968 in the name of Dale M. Uetrecht.

While we have described and illustrated one specific embodiment of our invention, it will be clear that variations of the details of construction which are specifically illustrated and described may be resorted to without departing from the true spirit and scope of the invention as defined in the appended claims.

We claim:

1. A system for generating chorus effects in music, comprising: control means for selectively placing said system in each

of at least first, second and third operating modes; a source of an audio spectrum; first means responsive to said system in said first operating mode for shifting by a first frequency deviation $f_1$ all the frequencies of said audio spectrum to form a processed audio band and responsive to said system in said second operating mode for shifting by a second frequency deviation $f_2$ all the frequencies of said audio spectrum to form said processed audio band; second means responsive to said system in said third operating mode for passing all the frequencies of said audio spectrum unshifted to form said processed audio band; means for maintaining the acoustic level of said processed audio band substantially equal for all said operating modes; and means for acoustically radiating said processed audio band, including: means for separating said audio spectrum into three components of equal amplitude and mutually spaced in phase by 120°; and means for applying said three components to said first means; wherein said first means includes: a three-phase subsonic oscillator for providing three oscillatory signals of equal frequency mutually phase separated by 120°; amplitude-modulation means for amplitude-modulating each of said three components with a respective one of said three oscillatory signals to provide three amplitude-modulated signals; and means for algebraically summing the three amplitude-modulated signals to provide said processed audio band, wherein said means for separating comprises: first phase-shift means for receiving said audio spectrum and providing one of said three components; second phase-shift means for receiving said audio spectrum and providing a second of said three components; means for algebraically summing said one and said second components; and means for inverting the sum of said one and second components to provide the third of said three components.

2. The system according to claim 1 wherein said first and second phase-shift means each comprises a second-order RC filter and a phase inverter connected in parallel therewith, wherein the term second-order filter means that the filter has a gain which is a second order function of the Laplace S operator.

3. The system according to claim 1 wherein said first and second phase-shift means each includes respective first and second second-order filter sections and a common phase inverter connected in parallel with both said filter sections, wherein the term second-order filter means that the filter has a gain which is a first order function of the Laplace S operator.

4. A system for generating chorus effects in music, comprising: control means for selectively placing said system in each of at least first, second and third operating modes; a source of an audio spectrum; first means responsive to said system in said first operating mode for shifting by a first frequency deviation $f_1$ all the frequencies of said audio spectrum to form a processed audio band and responsive to said system in said second operating mode for shifting by a second frequency deviation $f_2$ all the frequencies of said audio spectrum to form said processed audio band; second means responsive to said system in said third operating mode for passing all the frequencies of said audio spectrum unshifted to form said processed audio band; means for maintaining the acoustic level of said processed audio band substantially equal for all said operating modes; and means for acoustically radiating said processed audio band, including: means for separating said audio spectrum into three components of equal amplitude and mutually spaced in phase by 120°; and means for applying said three components to said first means; wherein said first means includes: a three-phase subsonic oscillator for providing three oscillatory signals of equal frequency mutually phase separated by 120°; amplitude-modulation means for amplitude-modulating each of said three components with a respective one of said three oscillatory signals to provide three amplitude-modulated signals; and means for algebraically summing the three amplitude-modulated signals to provide said processed audio band, wherein said amplitude-modulation means comprises: three DC differential amplifiers, one for each of said three components, each amplifier including one input terminal for receiving a respective one of said components and a second input terminal connected to AC ground; and means for varying the output current of each of said amplifiers at the frequency of a respective one of said three oscillatory signals.

5. A system for generating chorus effects in music, comprising: a source of an audio spectrum; control means for selectively placing said system in each of at least first, second and third independent operating modes; first means responsive to said system in said first operating mode for shifting by a first frequency deviation $f_1$ all the frequencies of said audio spectrum to form a first processed audio band, responsive to said system in said second operating mode for shifting by a second frequency deviation $f_1'$ all the frequencies of said audio spectrum to form said first processed audio band, and responsive to said system in said third operating mode for passing all the frequencies of said audio spectrum unshifted to form said first processed audio band; second means responsive to said system in said first operating mode for shifting by a first frequency deviation $f_2$ all the frequencies of said audio spectrum to form a second processed audio band, responsive to said system in said second operating mode for shifting by a second frequency deviation $f_2'$ all the frequencies of said audio spectrum to form said second processed audio band, and responsive to said system in said third operating mode for passing all the frequencies of said audio spectrum unshifted to form said second processed audio band; third means responsive to said system in said first operating mode for shifting by a first frequency deviation $f_3$ all the frequencies of said audio spectrum to form a third processed audio band, responsive to said system in said second operating mode for shifting by a second frequency deviation $f_3'$ all the frequencies of said audio spectrum to form said third processed audio band, and responsive to said system in said third operating mode for passing all the frequencies of said audio spectrum unshifted to form said third processed audio band; fourth means responsive to said system in said first operating mode for shifting by a first frequency deviation $f_4$ all the frequencies of said audio spectrum to form a fourth processed audio band, responsive to said system in said second operating mode for shifting by a second frequency deviation $f_4'$ all the frequencies of said audio spectrum to form said fourth processed audio band, and responsive to said system in said third operating mode for passing all the frequencies of said audio spectrum unshifted to form said fourth processed audio band; means for filtering said first, second, third and fourth processed audio bands to provide respective first, second, third and fourth audio subbands, each audio subband consisting of a different range of frequencies in said audio spectrum such that all frequencies of said audio spectrum are included in at least one audio subband; means for separately acoustically transducing and radiating each of said audio subbands, including: means for separating said audio spectrum into three components of equal amplitude and mutually spaced in phase by 120°; and means for applying said three components to each of said first, second, third and fourth means; wherein each of said first, second, third and fourth means includes: a three-phase subsonic oscillator for providing three oscillatory signals of equal frequency but mutually spaced 120° in phase; means for amplitude-modulating each of said three components with a respective one of said three oscillatory signals to provide three amplitude-modulated signals; means for algebraically summing the three amplitude modulated signals to provide said first, second, third and fourth processed audio bands, respectively, wherein said first, second, third and fourth means each includes: means responsive to said system in said first operating mode for providing said three oscillatory signals at frequency $f_1$ at said first means, $f_2$ at said second means, $f_3$ at said third means, and $f_4$ at said fourth means; and means responsive to said system in said second operating mode for providing said three oscillatory signals at frequency $f_1'$ at said first means, $f_2'$ at said second means, $f_3'$ at said third means, and $f_4'$ at said fourth means, wherein said first, second, third and fourth means include

means responsive to said system in said system in said third operating mode for passing only one of said three components, without modulation, as said first, second, third, and fourth processed audio bands, respectively, said only one component being of the same phase in each of said first, second, third and fourth means, and further comprising: means for amplifying said audio spectrum prior to separation into said three components by said means for separating; and means responsive to said system in said third operating mode for adjusting the gain of said amplifier means to compensate for gain variation resulting from the utilization of only one of said three components.

6. A circuit for separating an audio spectrum into three components of equal amplitude and frequency content but mutually spaced by 120° in phase, said circuit including: first second-order filter means for receiving said audio spectrum and providing one of said three components; second second-order filter means for receiving said audio spectrum and providing a second of said three components; means for providing an intermediate signal having an amplitude corresponding to the sum of the amplitudes of said one and said second components; and means for inverting said intermediate signal to provide the third of said components, wherein the order of a filter is the order in terms of the Leplace S operator which defines the gain of the filter.

7. The combination according to claim 6, including a common phase inverter connected in series with said first second-order filter and in parallel with said second second-order filter, wherein each of said first and second second-order filters includes: a first resistor, a first capacitor and a second resistor, all connected in series; and a second capacitor connected to ground from a junction between said first capacitor and said second resistor, wherein each of said first and second second-order filters additionally includes a third resistor, said third resistor for said first second-order filter being connected in parallel with the series-connected combination of said phase inverter and said first resistor, first capacitor and second resistor of said first second-order filter, said third resistor of said second second-order filter being connected in series across the series connected first resistor, first capacitor and second resistor of said second second-order filter, wherein the component values of said resistors and capacitors are selected to provide a difference between the phase versus frequency characteristics of said first and second second-order filters of substantially 120° over at least four octaves of said audio spectrum.

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