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3,476,863 11/1969 Campbell 84/1.01

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[54] **MUSICAL INSTRUMENT ELECTRONIC TONE
PROCESSING SYSTEM**
13 Claims, 3 Drawing Figs.

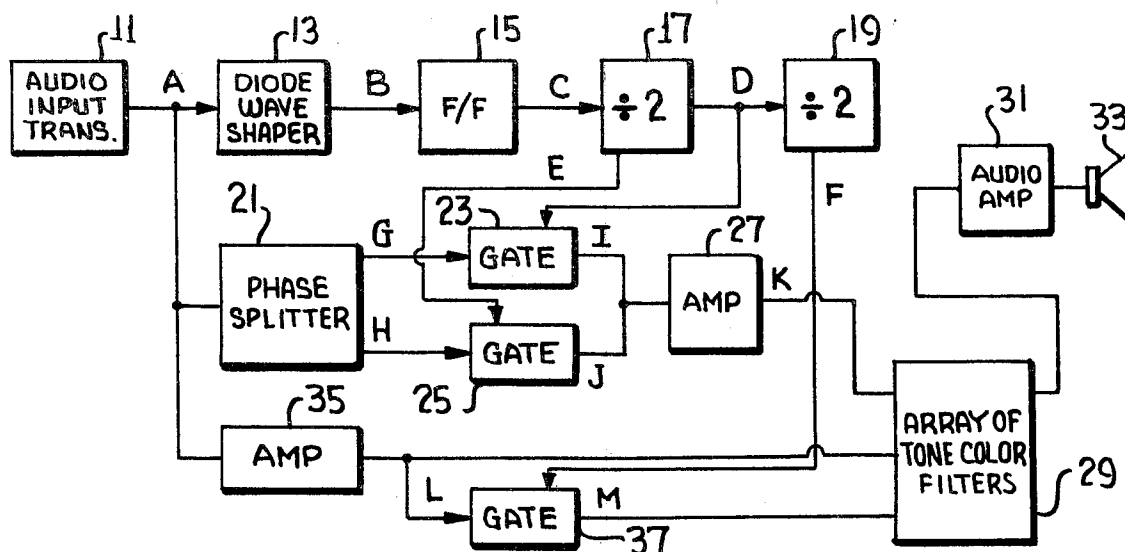
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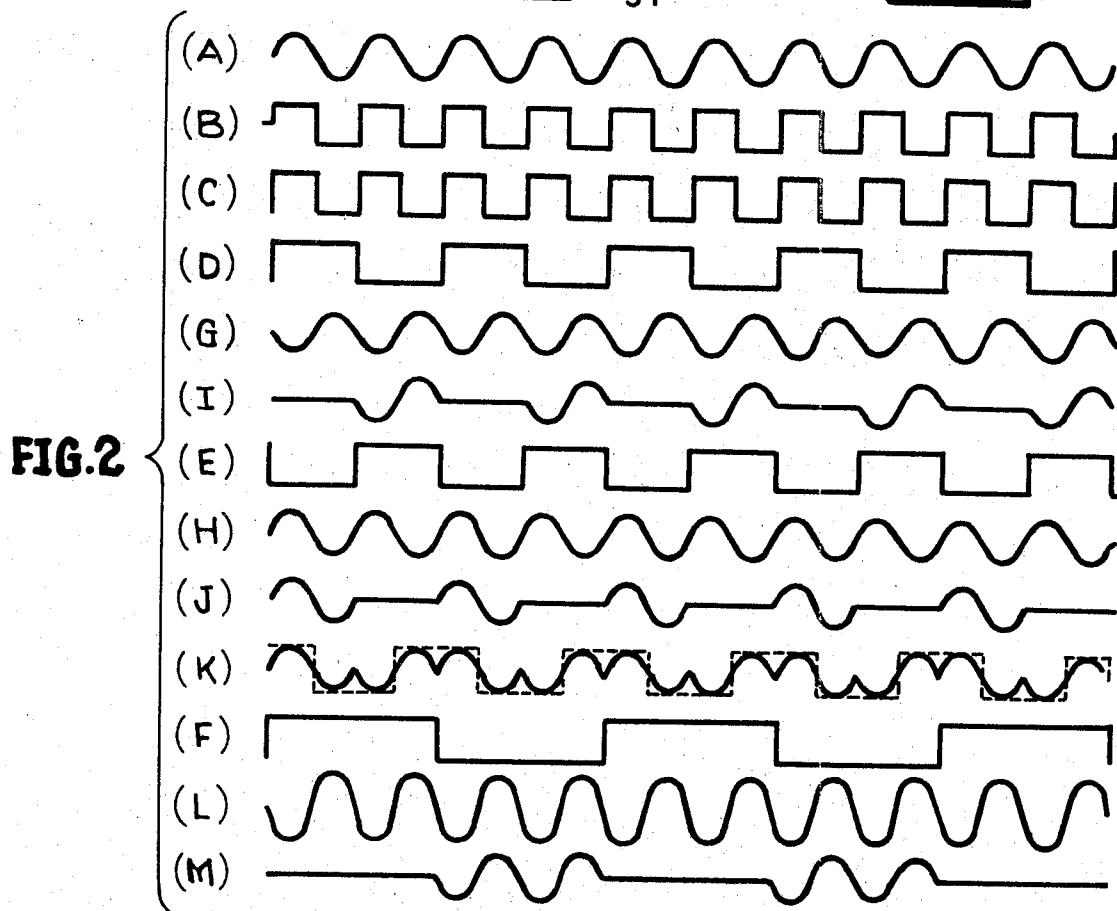
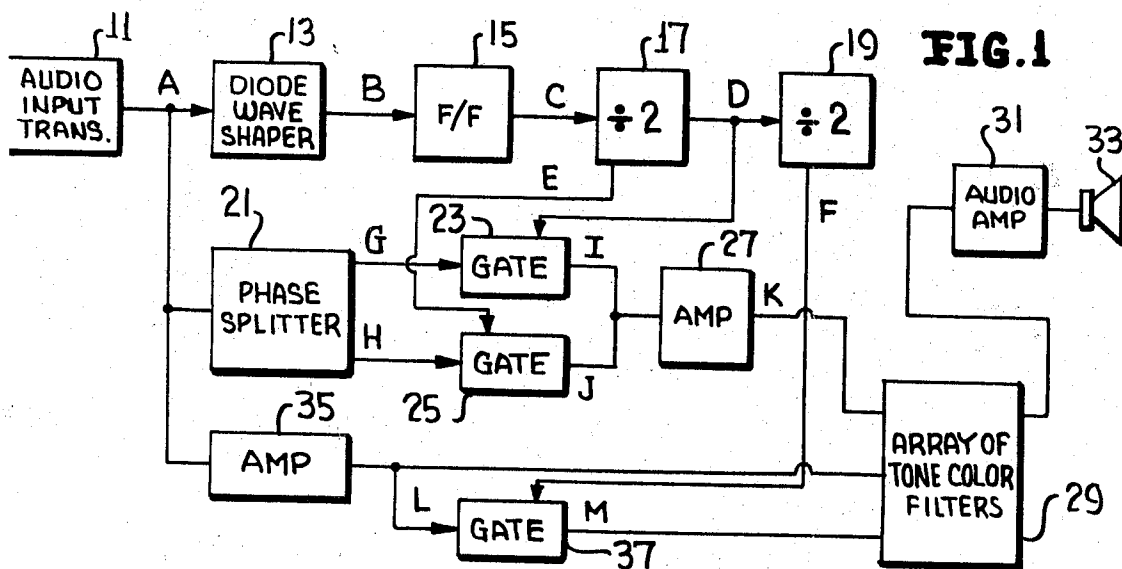
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ABSTRACT: An audio tone from a musical instrument is applied to a phase splitter to provide two oppositely-phased tone signals, each of which is passed to a respective field effect transistor transmission gate. In addition, the audio tone is converted to a square wave having the same frequency as the fundamental frequency of the tone. The square wave is passed to a frequency divider which provides a pair of gating signals for the respective transmission gates, the gating signals being oppositely-phased and at half the fundamental frequency of the tone. The oppositely-phased tone signals are alternately passed by the gates and combined in a tone color filter circuit which imparts specified musical tone qualities to the combined signal. In addition, the original tone may be gated via a further field effect transistor gate by a signal having a frequency which is one-quarter of the tone fundamental frequency, the gated signal being passed to an appropriate tone color filter. The original tone is also passed directly to a tone color filter. All of the filtered signals are then amplified and passed to a loudspeaker system to provide an acoustic signal of substantially greater tonal complexity than the original tone and which is controlled in frequency and amplitude by the frequency and amplitude respectively in the input tone.





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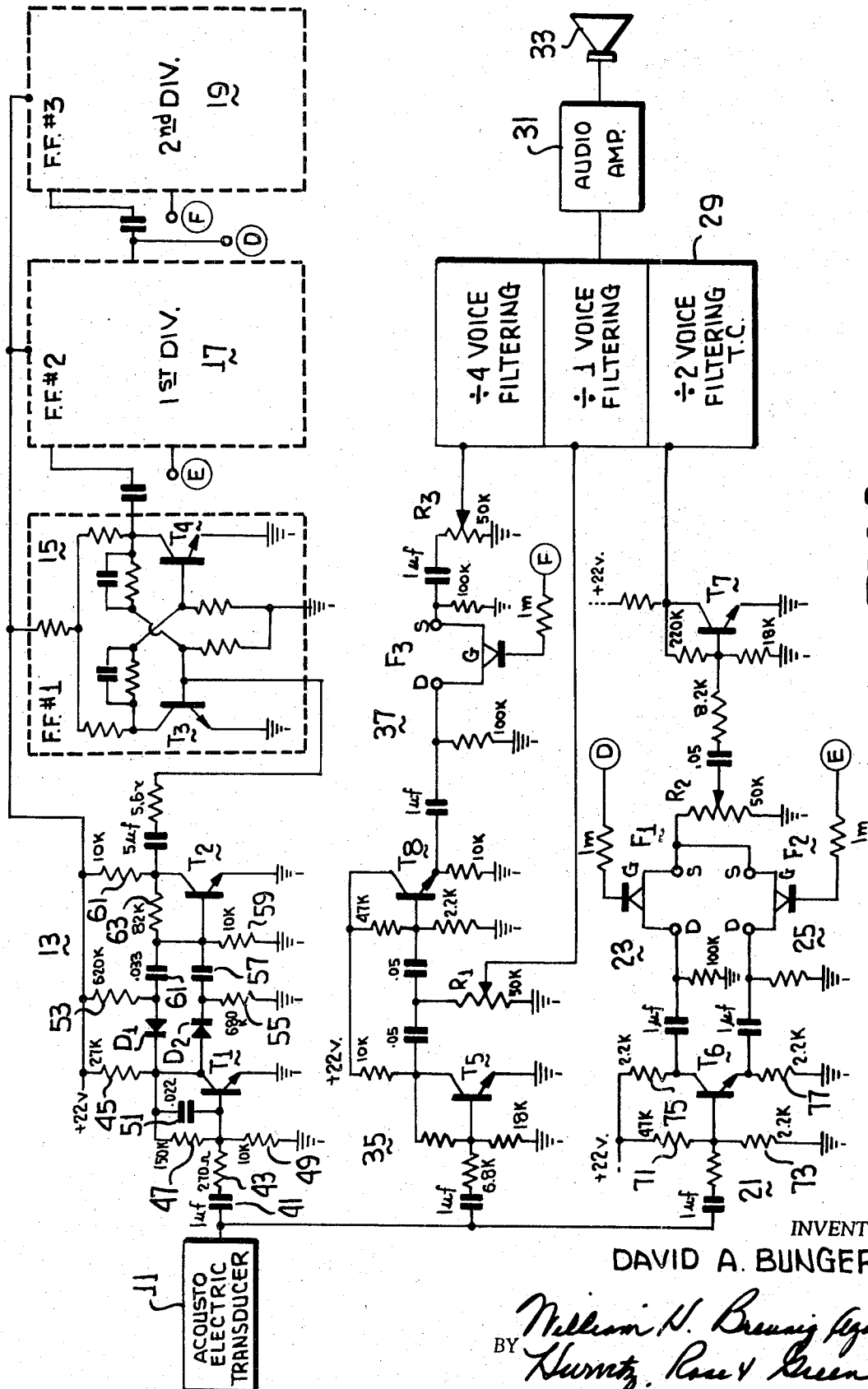


FIG. 3

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MUSICAL INSTRUMENT ELECTRONIC TONE PROCESSING SYSTEM

BACKGROUND OF THE INVENTION

The present invention relates to tone generation systems and more particularly to electronic systems which respond to acoustic tones from conventional musical instruments to provide signals of complex tonal quality.

The concept of electronically processing tones provided by conventional electric or nonelectrical musical instruments is known in the prior art. Examples of prior art approaches to such processing may be found in the following U.S. Pats.: Earp, No. 2,561,349; White, No. 3,006,228; Hanert, No. 2,514,490; and Cookerly et al. No. 3,213,180. My copending U.S. Pat. application, Ser. No. 712,117, filed Mar. 11, 1968 and entitled "TONE PROCESSING SYSTEM" discloses yet another approach to electronic processing of tones provided by conventional musical instruments. As discussed in said application, prior art systems lack a suitable device for assuring that the fundamental frequency of a complex musical tone will, for all frequencies and waveshapes, assume reliable control over the system. It is desirable, for example, that the simulated voices, electronically produced by the processing system in response to the input tone, have a signal level which follows changes in the input tone signal level as closely as possible. In addition, attack and decay characteristics of the artificial voices should be as realistic as possible, and this requires faster response to input tone level changes than was achievable in the prior art. Moreover, if the harmonic content of the input waveform for the input tone is changed, for example, by overblowing a clarinet, a corresponding tone color change should occur in the simulated voices.

My above referenced U.S. Pat. application discloses a system in which these characteristics are considered. More particularly, my prior system provides for conversion of the input tone to square waves having the frequency of the input tone fundamental frequency. The square wave is then distributed to frequency division and multiplication circuitry to provide signals corresponding to the various desired simulated voices. The different voice signals are then passed to respective tone color filter networks, combined, and then passed to a diode voltage controlled expression circuit which is responsive to level changes in the input tone to vary the level and expression characteristics of the artificially generated voice signals. The present invention is concerned with the same problem as was my prior system; however, the system of the present invention represents a significant improvement in providing greater control over signal level and expression of the simulated voices by the input tone that was achievable in prior art systems.

It is therefore an object of the present invention to provide an electronic processing system for input tones, no matter how generated, in which complex output tones are produced at frequencies and amplitudes which are controlled directly by the frequency and amplitude of the input tone.

It is another object of the present invention to provide a tone processing system of the type referred to above wherein output tone expression follows input tone variations more closely than in prior art tone processing systems.

SUMMARY OF THE INVENTION

In accordance with the principles of the present invention, the input tone signal itself, rather than square waves derived therefrom, is passed to the tone color filtering network via field effect transistor transmission gates which are operated at frequencies corresponding to the desired simulated voices to be electronically produced. One set of voice signals is produced by applying the input tone signal to a phase splitter which in turn applies a pair of oppositely-phased output signals to a respective pair of field effect transistor gates. The gating signals for controlling transmission through the gates are derived from the input tone signal by means of a pulse converter, which converts the input tone signal to a train of pulses

having a frequency equal to the fundamental frequency of the input tone signal, and a frequency divider which divides the pulse train frequency in half to produce two oppositely-phased gating signals each having a frequency equal to half the fundamental frequency of the input tone signal. Opposite phasing of the gating signal provides alternate operation of the two field effect transistor gates, the output signals from which are combined to provide a harmonically rich simulated voice signal at a frequency equal to one half the fundamental frequency of the input tone signal.

The input tone signal is also applied to an amplifier which applies the amplified version of the input tone signal to a further field effect transistor gate. The further gate is controlled by a gating signal having a frequency equal to one-quarter of the fundamental frequency of the input tone signal, this latter gating signal being derived from further frequency division of the pulse train derived from the input tone signal. The signal thus passed by the further transmission gate has a frequency equal to one-quarter of the fundamental frequency of the input tone signal and thereby represents another simulated voice produced electronically by the tone processing system of the present invention. The artificially generated voices, as well as the input signal itself are applied to an array of tone color filters which imparts desired musical characteristics and tonal qualities to the individual voices and in turn applied the resultant signals to an amplifier and loudspeaker system.

By utilizing the input tone signal itself, rather than square waves derived therefrom, to provide the basis of the individual simulated voices, the system provides a complex tonal output having a signal level and tonal expression which accurately follows the input tone signal level. In addition, the use of field effect transistors for transmission gates permits one to take advantage of the relatively noise-free characteristics of such transistors, whereby substantially no undesired frequency components are introduced in the simulated voice signals. Further, by alternately gating the oppositely-phased output signals from the phase splitter with correspondingly oppositely-phased gating signals having frequencies equal to one-half the fundamental frequency of the tone, a desirable wave form is obtained for producing realistic sound in the "divide-by-two" voices. More specifically, the combined portions of the gated phase splitter output signals tend to approximate, while not corresponding precisely to, a square wave having a frequency equal to one-half the input tone fundamental frequency. This type of wave form allows one to obtain very realistic bass clarinet and bassoon tones.

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 one specific embodiment thereof, especially when taken in conjunction with the accompanying drawings, wherein:

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

FIGS. 2A through 2M are plots of wave shapes of signals appearing at designated points in the system illustrated in FIG. 1;

FIG. 3 is a schematic diagram of a system according to FIG. 1.

DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring now to FIG. 1 of the accompanying drawings an audio input signal A, which may be derived from an acousto-electric transducer 11 operatively associated with a horn, clarinet, guitar, or the like, is applied to a diode waveshaper 13. By way of example and for facilitating description of system operation, the audio input signal may take the form of a pure sinusoid such as illustrated in FIG. 2A; however, it is to be understood that the present system is capable of processing input signals of substantially more complex waveshapes than a simple sinusoid. Signals appearing at different points in the system of FIG. 1 are designated A through M and are plotted

with respect to time in FIG. 2A through 2M respectively. Diode waveshaper 13 converts the input signal to a square wave or pulse train B as illustrated in FIG. 2B and which has a pulse repetition frequency (PRF) equal to the fundamental frequency of the audio input signal A. Pulse train output signal B drives a set-reset flip-flop 15 which in turn provides a pulse train C having a PRF equal to the audio input signal fundamental frequency, but a constant amplitude regardless of amplitude variations in audio input signal A. The wave shape of the output signal from flip-flop 15 is illustrated in FIG. 2C. The output signal from flip-flop 15 drives two cascaded "divide-by-two" circuits, 17 and 19, respectively. Circuit 17 provides two oppositely-phased square wave signals, D and E each having a frequency equal to one-half the fundamental frequency of audio input signal A, and which are utilized as gating signals in a manner to be described hereinbelow. Circuit 19 provides a square wave signal F which has a frequency equal to one-quarter the fundamental frequency of the audio input signal. Signals D, E and F are illustrated in FIGS. 2D, 2E and 2F, respectively, of the accompanying drawings.

Audio input signal A is also applied to a phase splitter network 21 which provides two oppositely-phased signals G and H of substantially equal amplitude and having a frequency equal to the fundamental frequency of input signal A. Signals G and H are illustrated in FIGS. 2G and 2H respectively of the accompanying drawings. Signal G, which is opposite in phase to signal A, is applied to the input terminal of signal transmission gate 23 and signal H, cophasal with signal A, is applied to the input terminal of signal transmission gate 25. Gate 23 also receives signal D at its gate terminal and gate 25 also receives signal E at its gate terminal. Each of gates 23 and 25 operate to pass signals applied to their input terminals in response to gating signals below a predetermined level at their gate terminals, and to block passage of signals applied to their input terminals in response to application of a signal above the predetermined signal level at their gate terminals. Gates which block signal transmission for low level gating signals and pass signals in response to high level gating signals may also be employed within the scope of the present invention. The signal passed by gate 23 in response to the gating action of signal D is designated as signal I and comprises alternate cycles of signal G. The signal passed by gate 25 in response to gating signal E is designated as signal J and comprises alternate cycles of signal H. Signals I and J are combined and applied to an amplifier 27 which provides an amplified version of the combined signals I and J designated as signal K. The effect of utilizing oppositely-phased gating signals D and E to gate respective oppositely-phased input signals G and H is to produce in signal K (see FIG. 2K) a signal corresponding to signal A but in which alternate cycles are phase-inverted by 180°. Where signal A is a sinusoid, signal K appears as a pair of adjacent negative half-cycles followed by a pair of adjacent and positive half-cycles, in iterative fashion. As indicated by the dotted line in FIG. 2K, signal K may thus be looked upon as a harmonically rich approximation of a square wave having a frequency equal to one-half the fundamental frequency of the audio input signal to the system. Where signal A is not a pure sinusoid, signal K is varied somewhat but remains a harmonically rich signal at half the fundamental frequency of signal A. Signal K comprises the "divide-by-two" voice derived by the tone processing system of the present invention and is applied to an array of tone color filters, selectable by operation of individual tabs, not illustrated. The tone color filters permit selective modification of the tone color of the signals applied thereto so as to produce as closely as possible the tones representing those of for example, the clarinet, oboe, flute, tuba, saxophone, or dulciana. The output signal from the array of tone color filters is applied to an audio amplifier 31, which in turn drives a loudspeaker 33.

The audio input signal A is also applied to amplifier 35, which introduces a 180° phase shift in the amplified signal L. A portion of signal L is applied directly to a section of the array of tone color filters 29 to comprise the "divide-by-one"

voice of the system. Another portion of signal L is applied to the input terminal of a transmission gate 37 which also receives at its gate terminal signal F provided by frequency divider circuit 19. Gate 37, like gates 23 and 25, passes signals from its input to its output terminals in response to gate signals below a predetermined level and blocks transmission of signals between its input and output terminals in response to gate signals above the predetermined level. The output signal from gate 37 is designated as signal M in FIG. 2M. Signal M is a chopped version of signal L, having a fundamental frequency equal to one-quarter of the frequency of the audio input signal to the system due to the fact that the frequency of gating signal F is one-quarter of said fundamental frequency. Signal M is applied to a further section of the array of tone color filters 29 and comprises the basis of the "divide-by-four" voice signal.

Referring now specifically to FIG. 3 of the accompanying drawings the audio input signal received from transducer 11 is applied via coupling capacitor 41 and base current limiting resistor 43 to the base of NPN transistor T1 connected to one end of a load resistor 45, the other end of which is connected to a +22v. DC supply. A voltage divider comprising resistors 47 and 49 is connected between the collector of transistor T1 and ground, and serves to bias the base of transistor T1. A capacitor 51 is connected between the base and collector of transistor T1 and provides frequency selective feedback to accentuate response of transistor T1 as an inverse function of frequency; that is, capacitor 51 provides rolloff which accentuates the fundamental frequency of a complex wave at the expense of the higher harmonics.

The amplified audio signal appearing at the collector of transistor T1 is superposed on a DC voltage of about 12v. representing the DC supply voltage of 22v. less the DC voltage drop across resistor 45. To the collector of the transistor T1 are directly connected the cathode of diode D1 and the anode of diode D2. A DC path exists through diodes D1 and D2 from the 22v. supply to ground through resistor 53 connected between the 22v. supply and the anode of diode D1 and resistor 55 connected between the cathode of diode D2 and ground. Nearly one-half of the available supply voltage is dropped across resistor 53, implying that the anode of diode D1 is at about 12v. and that both diodes are biased slightly conductive. When the audio signal goes positive it charges a capacitor 57 which is connected to the cathode of diode D2 at one end and to ground through resistor 59 at its other end. When the audio signal goes negative it charges capacitor 61 which is connected at one end to the anode of diode D1 and at its other end to ground through resistor 59. The audio signal when going positive, for example, charges capacitor 57 to approximately the peak level of the signal, and this charge is retained during a half cycle of the audio wave, so that if the positive half cycle is quite complex, as is usually the case, any dips of level in the course of the positive half cycle do not affect the output level at the base of transistor T2 to which the high voltage side of resistor 59 is connected. Similarly, when the audio signal goes negative with respect to the normal DC level at the collector of transistor T1, diode D1 charges capacitor 61 and prevents transient voltage drops during the next half cycle from generating reverse current flows through capacitor 61. Charging of capacitor 61 induces discharge of capacitor 57 and vice versa. Since transfer of the charging function between capacitors occurs as the input signal wave passes through zero level, the system can respond to AC waves over a wide range of frequencies, on the order of 50 Hz to 5,000 Hz. The double diode, double capacitor detection system, as first disclosed in my above-referenced U.S. Pat. application Ser. No. 712,117, may be denominated a push-pull peak detector, since it responds to alternate polarities of a wave to provide a clipped AC output signal which approximates a square wave. The closeness of the approximation depends, however, on the complexity of the input wave form; that is, the relative amplitudes and phases of its component partials.

Diode D1 and capacitor 61 is a peak detector for negative signals, and diode D2 and capacitor 57 is a peak detector for positive signals. The circuit therefore tends to produce a square wave at the peaks of the input waveform, changing state as the input signal reverses polarity. Current through the load resistor 59 therefore is an AC signal with respect to the DC bias for NPN transistor T2 established by resistors 61, 63 and 59. Resistor 61 is connected between the 22v. DC supply and the collector of transistor T2; resistor 63 is connected between the collector of transistor T2 and the base of transistor T2, and resistor 59 is connected between the base of transistor T2 and ground. Transistor T2 saturates upon sufficiently large positive signals occurring at its base, its output having values extending from zero to 22v. as the input AC waveform alternates, and it is designed to saturate on any expected level of output from transducer 11. The output of transistor T2 is therefore the square wave designated as signal B in FIG. 1 and illustrated in FIG. 2B.

The output signal from transistor T2 is AC coupled to set-reset flip-flop 15 which operates at the frequency of the input wave and comprises two cross connected NPN transistors T3 and T4 in a configuration which is conventional and hence is not described in detail herein. The output of flip-flop 15 is applied to cascaded flip-flops 17 and 19, each substantially identical to flip-flop 15, and arranged to operate as frequency dividers. Flip-flop 17 provides signals D and E and flip-flop 19 provides signal F.

The audio input signal A is also applied via a coupling capacitor and base current limiting resistor to the base of NPN transistor T5 which comprises amplifier 35 of FIG. 1. Transistor T5 is biased and loaded in a manner substantially similar to transistor T1 and provides at its collector an amplified, phase-inverted version of the audio input signal A. This amplified signal is coupled via an AC coupling capacitor to one end of a variable resistor R1 which has its slider arm connected to the "divide-by-one" voice filtering section of the array of tone color filters 29 so that resistor R1 provides selective adjustment of the "divide-by-one" voice level.

The amplified audio signal appearing across resistor R1 is applied to the base of NPN transistor T8 connected in emitter-follower configuration. The output signal appearing at the emitter of transistor T8 is AC coupled to the drain electrode of a P-channel field effect transistor F3, which corresponds to signal transmission gate 37 of FIG. 1. The source electrode of field effect transistor (FET) F3 is AC coupled to a variable resistor R3, having one end connected to ground and having its slider arm connected to the "divide-by-four" voice filtering section of the array of tone color filters 29. The gate electrode of FET F3 receives signal F from flip-flop 19.

The operation of FET F3 is as follows: When the gate electrode is referenced to a positive DC voltage higher than the pinch off voltage of the FET, the FET is off; that is, when the gate electrode is reverse biased there exists a relatively high impedance between the drain and source electrodes so that no signal is passed therebetween. When the gate electrode is grounded, the FET F3 is on and a very low impedance exists between the drain and source electrodes, and therefore a signal applied to the drain electrode is readily passed to the source electrode. Passage of a signal between the drain and source electrodes is accomplished with minimal attenuation, and the output signal from the FET is at a level which tracks the amplitude of the input signal thereto quite closely. The waveforms of the signals L, F and M appearing at the drain, gate and source electrodes respectively have been described in detail in reference to FIG. 1 above and are illustrated in FIGS. 2L, 2F and 2M respectively.

Audio input signal A is also applied via a suitable AC coupling capacitor and base current limiting resistor to the base of NPN transistor T6, which is connected in phase splitter configuration. More particularly, the base of transistor T6 is biased by a voltage divider network comprising a resistor 71 connected between a 22v. DC voltage source and the base of transistor T6, and resistor 73 connected between the base

of transistor T6 and ground. A resistor 75 is connected between the collector of transistor T6 and the 22v. source, and a resistor 77 is connected between the emitter of transistor T6 and ground. Resistors 75 and 77 are of equal value so that the signals provided at the collector and emitter respectively of transistor T6 are of substantially equal amplitude. However, the signal appearing at the collector of transistor T6 is a phase-inverted function of the AC signal applied to the base of transistor T6, whereas the signal appearing at the emitter electrode of transistor T6 is not phase-inverted. This phase-splitting action of transistor T6 provides oppositely-phased signals having frequencies which are equal to the frequency of audio input signal A and having amplitudes which closely track the amplitude of input signal A. The collector of transistor T6 is AC coupled to the drain electrode of field effect transistor F1 comprising gate 23 of FIG. 1. Similarly the emitter electrode of transistor T6 is AC coupled to the drain electrode of field effect transistor F2 comprising gate 25 of FIG. 1. Field effect transistors F1 and F2 receive signals D and E respectively at their gate electrodes, and operate in a manner as substantially described for transistor F3 above. Thus signal G appearing at the collector of transistor T6 is selectively gated through FET F1 by signal D to provide signal I, whereas signal H appearing at the emitter electrode of transistor T6 is selectively gated through FET F2 by signal E to provide signal J. Signals I and J are combined and applied across variable resistor R2 to ground, the center arm of which is AC coupled to the base of NPN transistor T7. Transistor T7 is connected in conventional class A amplifier configuration and provides signal K to the "divide-by-two" section of the array of tone color filters 29. The signal level of the "divide-by-two" voice signal is selectively adjustable by means of R2.

The output signals from the array of tone color filters 29 are combined and applied to audio amplifier 31 and in turn to loudspeaker 33 as described above in relation to FIG. 1.

In summary the various individual voices artificially obtained by the system of the present invention are produced as follows:

The "divide-by-one" voice signals are obtained by amplifying the original audio input signal A by means of amplifier 35 and applying the resulting wave form to the respective "divide-by-one" voice filtering section of the array 29;

The "divide-by-two" voices are obtained by feeding the original audio input signal A to phase splitter 21, the oppositely-phased output of which are selectively gated by oppositely-phased gating signals at one-half the fundamental frequency of the audio signal, the gated signals being combined and applied to the "divide-by-two" voice filtering sections of array 29;

The "divide-by-four" voice signals are obtained by gating audio signal A by a signal corresponding to one-quarter of the frequency of the input signal and applying the resulting gated signal to the "divide-by-four" voice filtering section of array 29.

The reason for employing phase splitter 21 to produce the "divide-by-two" voice signals, is that the waveform of signal K is desirable from a filtering standpoint for use in obtaining realistic sounds of "divide-by-two" voices. It can be seen, for example, that the waveform of signal K tends to approximate, without actually corresponding identically to, a square wave with a fundamental frequency equal to one-half the input fundamental frequency. This type of wave form has been found to be conducive to readily obtaining a realistic bass clarinet voice.

Utilization of the FET gates to selectively pass the audio input signals at gating frequencies which are submultiples of the fundamental signal frequency provides much more realistic attack and decay characteristics of the various simulated voice tones than was achievable with prior art systems. The FET gates themselves are relatively noise free, and therefore introduce substantially no undesired frequency components in the simulated voices of the processing system. In addition, by using the actual input tone for producing the vari-

ous voices, the output is expressed precisely as the input. If the harmonic content of the input waveform is changed, for example, by overblowing the instrument providing the input signal, the corresponding tone change occurs in the artificial voices with varying degrees. Tonal expression is achieved with much faster response to input signal level changes than prior art systems, and the various artificial voices passed through the FET gates are permitted to have level change ranges which correspond to the range of input signal level.

While I have described and illustrated one specific embodiment of my 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 claim.

I claim:

1. In a tone signal processing system for imparting complex tonal qualities to an input tone signal:

pulse converter means responsive to said input tone signal for providing a train of pulses having a fundamental frequency equal to the fundamental frequency of said input tone signal;

phase splitter means responsive to said input tone signal for providing a pair of oppositely-phased signals at the fundamental frequency of said input tone signal;

first gating means responsive to a first gating signal for selectively passing a first of said pair of oppositely phased signals;

second gating means responsive to a second gating signal for selectively passing a second of said pair of oppositely phased signals;

frequency divider means responsive to said train of pulses for providing as said first and second gating signals respective first and second gating pulse trains of opposite phase and at a frequency equal to the fundamental frequency of said input tone signal divided by two raised to a predetermined integer power; and

means for combining the portions of said oppositely-phased signals passed by said first and second gating means.

2. The system according to claim 1 further comprising tone color filter means connected to said means for combining for musically modifying the combined portions of said oppositely-phased signals passed by said first and second gating means.

3. The system according to claim 1 wherein said specified integer power of two is unity.

4. The system according to claim 1 wherein said first and second gating means each comprises a field effect transistor having source, drain and gate electrodes, said field effect transistors being connected for selectively passing signals between said source and drain electrodes in response to application of a gating signal to said gate electrode.

5. The system according to claim 1 further comprising: third gating means responsive to a third gating signal for selectively passing said input tone signal; and

wherein said frequency divider means includes means for providing as said third gating signal a pulse train having a frequency equal to the fundamental frequency of said input tone signal divided by two raised to a further integer power other than said specified integer power.

6. The system according to claim 5 wherein said specified integer power is one and said further integer power is two.

7. The system according to claim 6 further comprising tone color filter means for musically modifying the combined portions of said oppositely-phased signals passed by said first and second gating means and the portion of said input tone signal passed by said third gating means.

8. The system according to claim 7 further comprising additional tone color filter means for musically modifying said input tone signals independently of those portions of said input tone signal passed by said first, second and third gating means.

9. The system according to claim 7 wherein said first, second and third gating means each comprises a field effect transistor having drain source and gate electrodes, each said field effect transistor being connected to pass its respective signals between said drain and source electrodes in response to a respective gating signal at said gate electrode.

10. The system according to claim 9 wherein said pulse generator means comprises a push-pull peak rectifier circuit connected to said transducer means, said circuit comprising:

a first junction to which said electrical signal is applied; a first diode having its cathode connected to said first junction;

a first capacitor, one end of which is connected to the anode of said first diode;

a second diode having its anode connected to said first junction;

a second capacitor, one end of which is connected to the cathode of said second diode;

means for connecting the other ends of said first and second capacitors together at a second junction;

a load resistor connected between said second junction and ground;

a transistor clipping circuit responsive to the voltage across said load resistor and arranged to become alternately nonconductive and saturated as the voltage across said load resistance changes algebraic sign; and

a flip-flop having a single control terminal responsive to said transistor clipping circuit.

11. In a system for processing the variable frequency acoustic tone of a nonelectrical musical instrument;

a transducer for coupling said acoustic tone to said system and converting said tone to an electrical signal having a frequency equal to the acoustic tone frequency and an amplitude proportional to the acoustic tone amplitude;

gating means responsive to a gating signal for selectively passing said electrical signal, said gating means comprising a field effect transistor having drain, source and gate electrodes, said electrical signal being selectively passed between said drain and source electrodes in response to application of said gating signal to said gate electrode;

pulse generator means responsive to said electrical signal for providing a train of pulses at a frequency equal to the fundamental frequency of said acoustic tone;

frequency divider means responsive to said train of pulses for providing a further train of pulses having a frequency equal to the frequency of said first mentioned train of pulses divided by some integer power of two; and

means for applying said further train of pulses to said gate electrode as said gating signal.

12. In the system according to claim 11, tone color filter means for musically modifying the electrical signal passed by said gating means.

13. The system according to claim 12 wherein said pulse generator means comprises a push-pull peak rectifier circuit connected to said transducer means, said circuit comprising:

a first junction to which said electrical signal is applied; a first diode having its cathode connected to said first junction;

a first capacitor, one end of which is connected to the anode of said first diode;

a second diode having its anode connected to said first junction;

a second capacitor, one end of which is connected to the cathode of said second diode;

means for connecting the other ends of said first and second capacitors together at a second junction;

a load resistor connected between said second junction and ground;

a transistor clipping circuit responsive to the voltage across said load resistor and arranged to become alternately nonconductive and saturated as the voltage across the load resistance changes algebraic sign; and

a flip-flop having a single control terminal responsive to said transistor clipping circuit.