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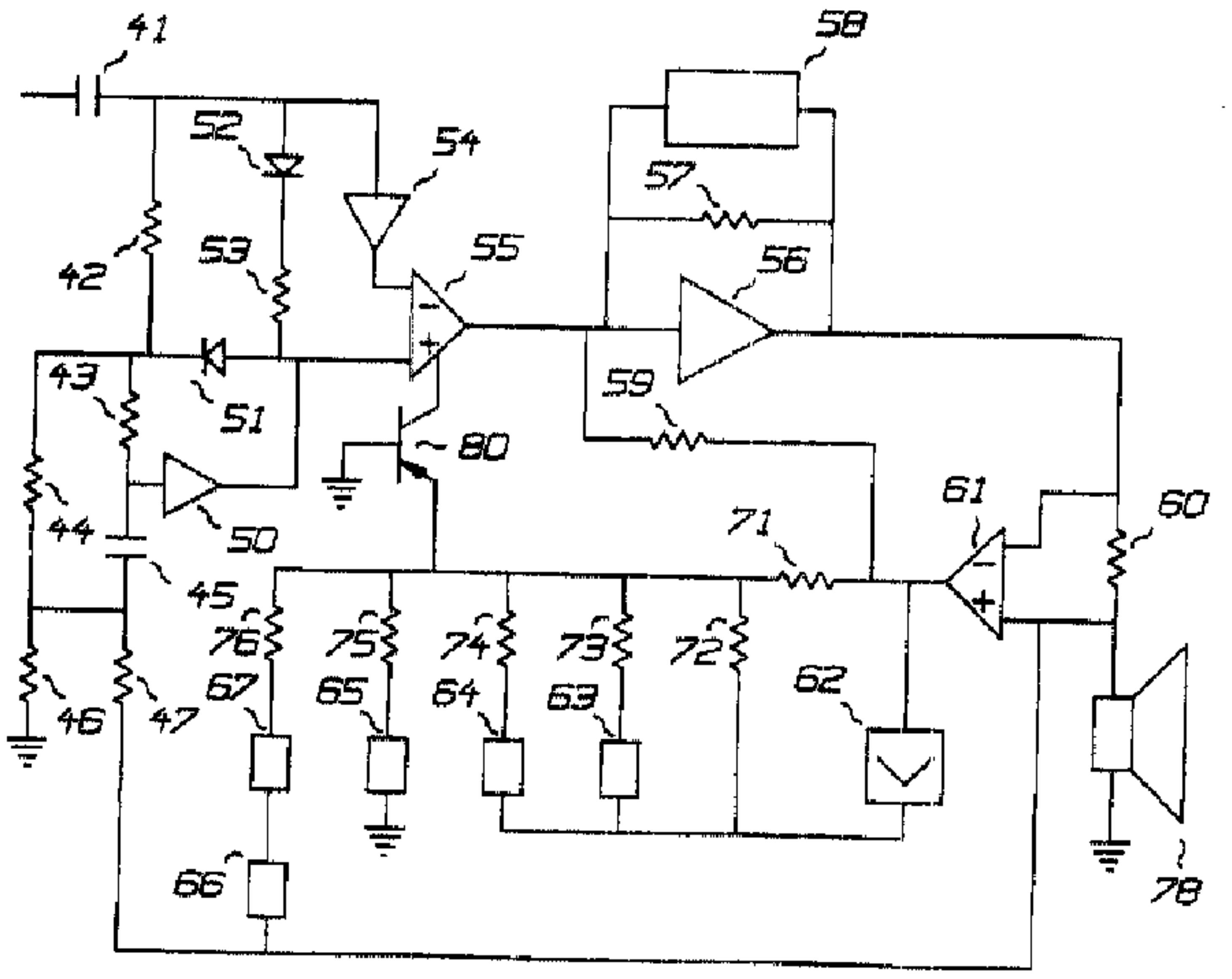

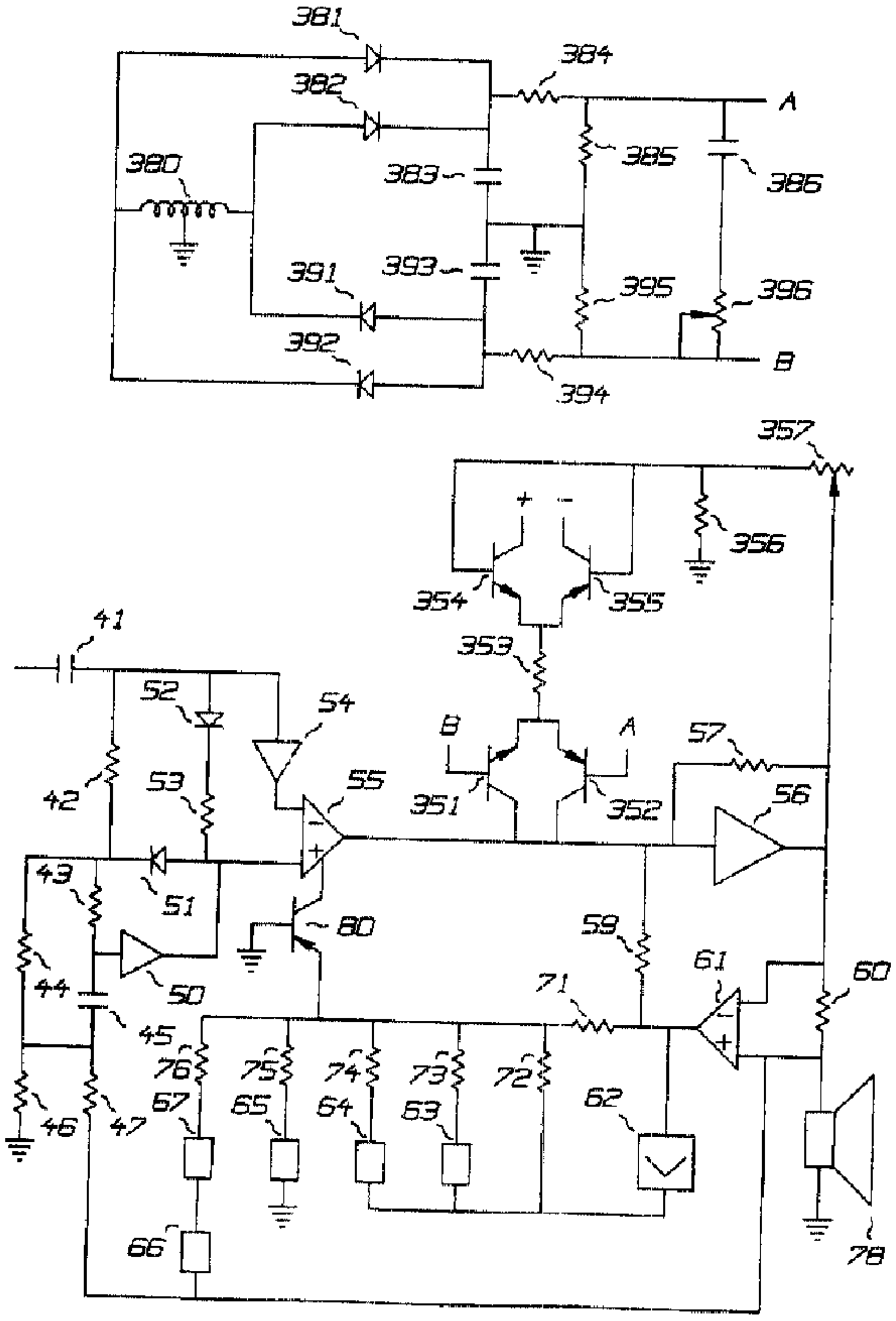

(57) **Abrégé/Abstract**:
This discloses various intermodulation means for the emulation and exaggeration of an aspect of vacuum tube amplifiers by solid state, digital, speaker, or other means. The intermodulation means produces intermodulation products of the input (2) and an audio signal source (4) which may be a spectrum-limited, filtered version of the input or the output or a spectrum-limited repetitive or random noise generator (8).

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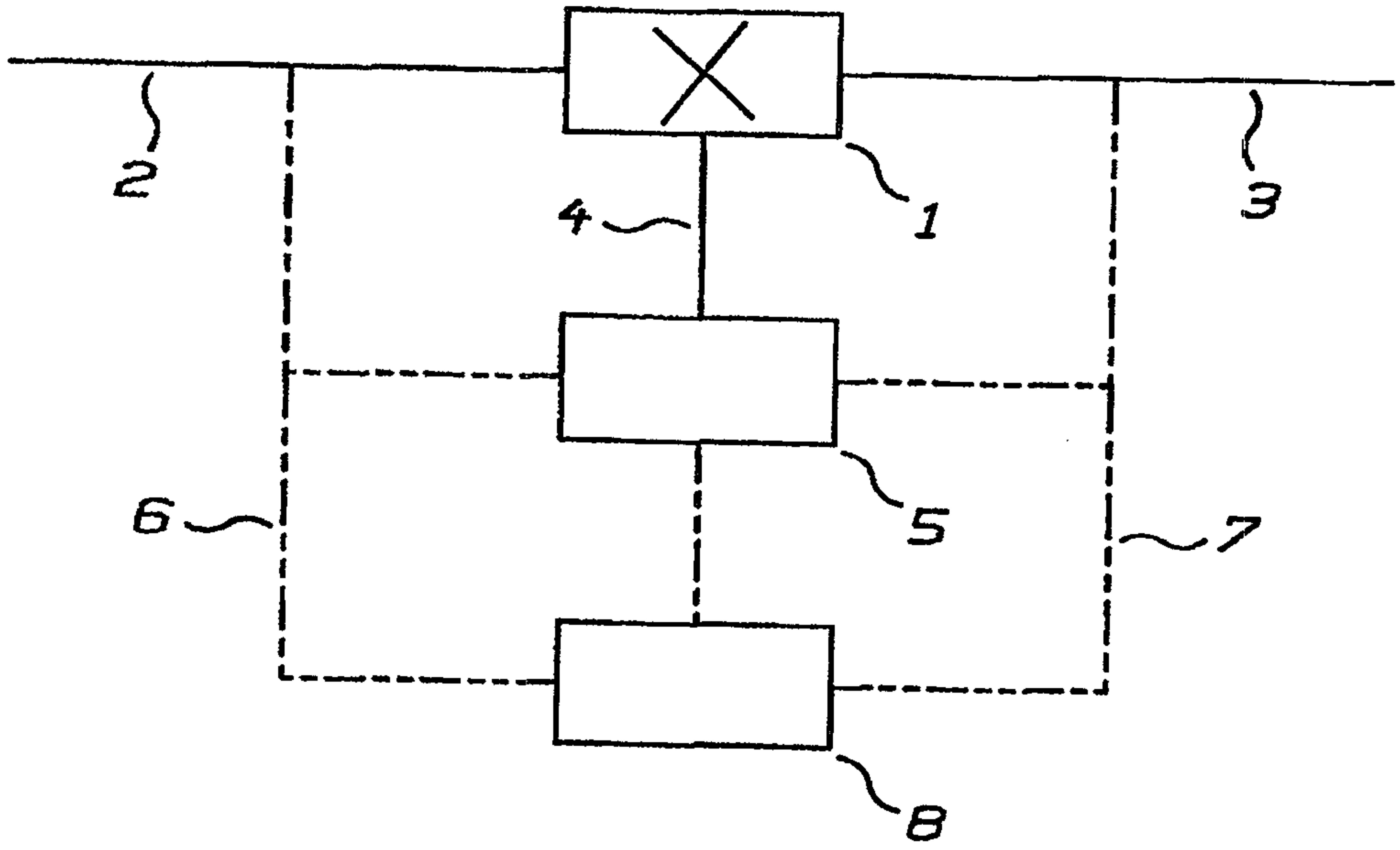

(57) Abstract

This discloses various intermodulation means for the emulation and exaggeration of an aspect of vacuum tube amplifiers by solid state, digital, speaker, or other means. The intermodulation means produces intermodulation products of the input (2) and an audio signal source (4) which may be a spectrum-limited, filtered version of the input or the output or a spectrum-limited repetitive or random noise generator (8).

FAT SOUND CREATION MEANS

BACKGROUND AND SUMMARY OF THE INVENTION

The present invention relates to the emulation of tube amplifiers, more particularly to the emulation of the fat created by intermodulation distortion, and extends the fundamental mathematics to the structure of speakers and computer programming for audio. The present invention provides audio effects that are so mysterious to at least the guitar amplifier industry that the expert amplifier writers have not published anything dealing with the phenomenon created by intermodulation distortions.

Although it has not been realized, the prior art is the power stage of the vacuum tube amplifier. There is an intermodulation of the power supply ripple with the input signal created by any or all of the comparatively low output resistance of vacuum tubes or the reaction to the ripple on the bias supplies by the screen or control grids. The engineering community has not found this character desirable because it violates their basic paradigm that amplifiers must replicate their inputs without embellishments. However, the more artistic appreciate these embellishments although they do not know their source.

Also similar, but inadequate, art is the tremolo circuit used by many older guitar amplifiers. However, the spectrum of the tremolo signal is too low to be noticed as a note because it is slow enough to create perceptible level changes. The -3 db point of the tremolo signal spectrum is far below 50 Hertz.

More distantly related art is the audio compressor, for example Scholz, U.S. 4627094. It measures the input or output signals and changes its gain to produce a less dynamic output signal. The measurement of the input or output signals is characterized by a rectification means, low-pass filtering means, and by a D.C. component that is responsive to the input signal. Since the compressor is not supposed to produce harmonics, intermodulation produces, or other embellishments when the input is constant, said filter signal is further characterized by having no audible signal for a constant input. This is not the intent of the present invention.

Additionally, Knopple, U.S. 4150253, distorts the output of a high-pass filter and adds the result with the original signal. And Liljeryd, U.S. 4731852, uses a constant 90 degree phase shifter and multiplier to produce only the sum frequency intermodulation products.

The speaker structure art has multiple winding speakers. Both Miessner, U.S. 1830402, and Bussard, U.S. 19777469, depend upon the power supplies of vacuum tubes to power the field coil. These speakers became obsolete about 45 years ago with the production of the permanent magnet speaker. This speaker is so cost ineffective that amplifier systems used the permanent magnet speaker in spite of having to provide the power supply with a filter choke. Consequently, the only reason for using this type of speaker is for its heretofore unknown special character, the intermodulation of the field coil with the voice coil.

Another speaker without a permanent magnet is Dinh, U.S. 5487114, which operates with a field coil that is connected to the input via a bridge rectifier. The consequential D.C. current in the field of Dinh is dependent upon the input and is filtered by the inductance of the field coil. Unfortunately, it does not work at low levels and requires extra power. The extra power requirement would probably adversely affect the tone of a guitar amplifier.

A high-fidelity speaker was disclosed by Lokkesmoe in U.S. 2727949 that included a permanent magnet as well as a field coil. The field coil and its parallel connected 25-30 Hertz band-pass filter extended the frequency response of the speaker. The extension of the frequency response by the field coil would require a significant power. This is consistent with a further analysis of the Lokkesmoe speaker. The series connected capacitor 22 is chosen to resonate the field coil and other connected inductors at 25-30 Hertz or about 175 radians per second. Since it is resonant, it probably has a Q of about 1. This forces the R/L frequency to be 175. A speaker of that era was modeled in *Radiotron Designer's Handbook*, 4th edition, 1953, page 838, with a voice coil inductance of 2.4 millihenries and D.C. resistance of 10.4 ohms. Since the Lokkesmoe design uses a field coil "preferably of higher" inductance than the voice coil, the field coil might be 10 millihenries. This implies a field coil and other connected inductors have a total D.C. resistance of 1.75 ohms. Thus, the field coil will draw more power than the voice coil. Although, one might believe the field coil inductance might reduce its power drain at higher frequencies, the magnetic losses at higher frequencies probably keep the power requirements up.

Moog, U.S. 4180707, has a multiplicative means driven by an input and a high-pass filter that does not restrict the upper audio spectrum.

Radiotron Designer's Handbook, pp 1322-1323, edited by F. Langford-Smith, 1953, RCA Victor Division, Radio Corporation of America, the only reference to mention intermodulation, describes hum distortion as the intermodulation of the power supply frequencies with the input signal at high volumes due to undersized power supply capacitors. This reference indicates that hum distortion is often overlooked when dealing with individual sources of distortion and, in fact, hum distortion was not included in the distortion section of this handbook. Also, hum distortion was not regarded as desirable.

The control grid bias supply of a vacuum tube amplifier is a potential source of an intermodulation signal source. However, it has not been a source because it has always been too easy to follow the engineering ideal of having essentially no ripple. Further, most amplifiers only use a half-wave rectification which has not been identified with goals of the present invention.

OBJECT OF THE INVENTION

The object of this invention is the intermodulation embellishment of an audio input signal with low-frequency, upper spectrum limited audio signal which does not include rectification and filtering of the input or the output and which is not the power supply of a tube amplifier. Further objects of this invention are the specific application of this concept to speakers, speaker emulators, clipping means, and amplifiers.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is the block diagram of the theory.

Figure 2 is the speaker embodiment.

Figure 3 is a speaker emulator embodiment.

Figure 4 is a first amplifier embodiment.

Figure 4A is the first amplifier embodiment modified for ripple modulation upon clipping.

Figures 5 and 5A are power supply embodiments.

Figure 6 is a controlled generator embodiment.

Figure 7 is a controlled bandwidth random noise embodiment.

Figure 8 is a parallel resistor-diode non-linear network.

Figure 9 is a series resistor-diode non-linear network.

Figure 10 is a diode-transistor non-linear network.

Figure 11 is a symbol for a non-linear network.

Figure 12 is a second amplifier embodiment.

Figure 13 is a third amplifier, variable resistance embodiment.

Figure 14 is the digital embodiment.

Figure 15 is a computer program flow chart.

DETAILED DESCRIPTION OF THE FIGURES

Figure 1 shows a non-linear means 1, such as a multiplying means, receiving an input on signal path 2 and producing an output 3. The non-linear means has a second input 4 which is created by a low-frequency means 5. The low-frequency means produces a signal composed of an audible low-frequency audio signal with a fundamental generally below 1000 Hertz or a spectrum which is more limited in the high frequencies than the input spectrum on signal path 2. Additionally, the spectrum below 50 Hertz has little use in guitar applications. This low frequency signal may be created by a signal source independent of the input, such as a generator or a power supply, or may be created by a low-frequency audio filter which is dependent upon either the input or the output, as shown by signal paths 6 and 7. This filter, to keep the spectrum limited to low-frequencies relative to the spectrum of the input, is a low-pass or band-pass with a resonant frequency or roll-off frequency above 50 Hertz and below 1000 Hertz or a fraction of the input spectrum.

The non-linear means creates intermodulation products of the signals on paths 2 and 4. The signal on path 2 and the D.C. component of the signal on path 4 combine to replicate the input signal on the output. The signal on path 2 and the audio component of the signal on path 4 combine to embellish the output intermodulation products of the two signals. In the case of the low-frequency source being dependent upon the input or the output, those intermodulation products can include harmonics of the input signal. The non-linear means creates intermodulation products whether or not the non-linear means is clipping.

The low-frequency means 5 may also be a controlled generator such as a voltage controlled oscillator. This provides an opportunity to match the spectrum of the generator with the spectrum of the input or output, generally keeping the spectrum of the generator a fraction of the input. This is accomplished by providing a control input which is frequency dependent. The signal on signal path 6 or 7 is filtered with a high-pass filter and then rectified to provide said control signal, see Figure 5.

Although there are many broad spectrum oscillators, the preferred oscillator has a waveform between a triangle and a sawtooth. The triangle wave form has every odd harmonic with amplitudes that roll off at 12 db per octave like a two pole filter. The sawtooth wave form has every harmonic and the amplitudes roll off at 6 db per octave, quite similar to a single pole low-pass filter or a low-Q band-pass filter. In either case the spectrum of a low-frequency means is limited and in contrast to a high-pass filter which has an unlimited spectrum above the frequencies of interest.

Empirically, the oscillator should produce frequencies in the 100 to 300 Hertz range.

The low-frequency means 5 may be a band-limited random noise generator. The use of a random noise generator provides a generally unrecognizable signal instead of the well-known power supply hum or the readily identified oscillator signal. Although these signals are not heard at low levels, they can be heard at high levels. The random noise generator still fattens the notes but does not produce an extra recognizable signal.

This concept is advanced by moving the band-limited random noise generator to 8 and placing a variable bandwidth filter between in 5. The variable bandwidth filter has a bandwidth dependent upon the frequency/amplitude of either the input or the output signals on paths 6 or 7, see Figure 6.

Additionally, Figure 1 shows yet another alternative path for dependency upon the input or output via the compressor 8. Since these effects are quite level dependent, the compressor serves to reduce the level dependency and spread the embellishment effect over a broader range of inputs. The compressor makes the controlled generator embodiment less level dependent.

Unlike the compressor, the signal path 4 has an audible signal even when the signal at the input is constant.

The signal injection, via path 4, needs to be subtle for, in this case, too much is not a good thing, yet contrary to accepted thought, none is not as good either. The use of the input or the output via paths 6 or 7 produces second harmonics which should be limited below 10 percent. The use of an oscillator should be more limited, to about 2 percent.

As shown in Figures 2 and 3, the non-linear means input which is connected to the input can include a series capacitor, such as 18 or 19, or other filter means to reduce the bass frequencies going through the non-linear means and consequentially the production of harmonics of those bass frequencies.

An important characteristic of this invention is that the signal on path 4 is not a significant component in the output 3. This is not the case when two signals are combined linearly and then distorted.

Thus, the output contains the signal components of the input signal and intermodulation products of said input and said limited spectrum source means and comparatively less of the signal from said limited spectrum source.

THE SPEAKER EMBODIMENT

Figure 2 is the speaker embodiment showing a permanent magnet 10 which produces a magnetic field that is conducted by an inner pole piece 11 and an outer pole piece 12 to the magnetic gap created for the voice coil 13. The voice coil drives the speaker cone 14. For clarity the remaining standard speaker components, frame and cone suspension, are not shown but are required.

The improvement to this speaker is the additional coil or the field coil 15 which is preferably wound on the inner pole piece 11.

This coil can be wound to have a significant inductance and resistance and thereby forms a low-frequency low-pass filter which may be augmented external components as well-known to the filter arts. Like the voice coil, this field coil is responsive to the input. It may be directly connected to the speaker terminals 17 or connected via a lamp 16. Additional filtering may be added to either connection. The resistance characteristic of a properly sized lamp produces little attenuation at low input signal levels, but a substantial attenuation at high input signal levels to extend the range of the embellishment.

The embellishment is formed by the interaction of the signal in the voice coil with the signal in the field winding. While the usually expected output is formed by the non-linear, approximately multiplicative, interaction of the signal and the permanent magnet, the embellishments are formed by the same non-linear, approximately multiplicative, interaction of the signal in the voice coil with the filtered signal in the field winding. The field coil can produce a signal in the output by inducing a current into the voice coil. However, this is not efficient and is comparatively less than driving the voice coil directly.

The speaker permanent magnet produces most of the magnetic field, substantially more field than the field winding. This magnetic field biases the field coil to produce a net field at the voice coil.

Unlike the prior art, Lokkesmoe U.S. 2727949, the field coil is intended to produce a moderate amount of intermodulaton distortion. The power required to produce an intermodulation distortion which enhances the sound instead of detracting from the sound is substantially lower than the apparent power requirements for extending the frequency response. Consequentially, the Lokkesmoe described coupling between the field coil and the voice coil does not produce any extension in the frequency response. Thus, the field coil of the present speaker invention falls into the pattern of this disclosure of producing intermodulation without adding significantly to the output.

The power requirements of the present invention field coil are substantially lower since the field coil of the present invention has a D.C. resistance higher than the voice coil. Although high fidelity speakers may have low efficiencies, low efficiency is not universally acceptable and particularly not acceptable for guitar speakers. Such a high resistance precludes series resonance at very low frequencies as found in the prior art.

For clarity, Figure 2 is not to scale. In reality, the magnet 10 is substantially thinner than shown and consequently minimizes the length of the field coil 15. Also, the ceramic magnet used today are thinner and the magnetic circuit is much shorter than the Alnico magnets used in the past because the ceramic magnet has a much higher coercive force. This makes the space available for the field coil much smaller. Further, as shown, the voice coil moves over the field coil and constrains its outer diameter. The inner diameter is also constrained by the desire to keep the reluctance of the magnetic field path low. Thus, the substantial field coil required by Lokkesmoe is not practical now.

The interaction of the voice coil with the permanent magnet produces the input signal. The interaction of the voice coil and the field coil produces intermodulation products. The field coil via other means produces comparatively less of the signal than the voice coil and the permanent magnet.

The upper minus 3 db roll-off point of the Lokkesmoe band-pass filtering for a resonant frequency of 30 Hertz and assuming a Q of 1 is about 48 Hertz, generally too low for successful operation according to the precepts of the present invention.

The speaker embodiment can also use a broad-spectrum, low-frequency oscillator to drive the field coil, however the transformer coupling from the field coil to the voice coil coupled with the finite impedance of the driving amplifier allows the oscillator to be heard, however, a third winding, co-located with the voice coil and field coil, such as found in Miessner serves to cancel oscillator signal, but not the intermodulation products. However, the Miessner speaker has been obsolete for about 45 years. It is more expensive to build and to use than the standard permanent magnet speaker.

The fat concept is also applicable to higher frequency speakers, such as tweeters. In this case capacitor 18 or other filter means is used to remove the bass frequencies in the voice coil. Thus, the harmonics of the low-frequencies passed to the field coil are eliminated, but many of the intermodulation distortion enhancements remain.

THE SPEAKER EMULATOR EMBODIMENT

Figure 3 is the speaker emulator embodiment that also shows the fundamentals of Figure 1. The input is received and attenuated by resistors 20 and 21 to drive a transconductance operational amplifier 22. Optional capacitor 19 also attenuates bass frequencies. The output current of this amplifier plus the additional current including bass frequencies from the input via resistor 23 drives a low-frequency filter created by components 24-28. The frequency of this filter is nominally the resonant of the frequency of the speaker being emulated. The combination of resistors 24 and 25, capacitor 26, and operational amplifier 27 appears to capacitor 28 to be a parallel combination of a resistor equivalent to the parallel combination of resistors 23-25 and an inductor equivalent to the product of the resistors 24 and 25 and capacitor 26. These equivalent components combine with capacitor 28 to create a resonant circuit. The output of operational amplifier 27 is then an underdamped low-pass filter which drives the bias input of the transconductance amplifier 22 via resistor 30. Since the bias input of the preferred transconductance amplifier, either a Harris CA3080 or National LM3080, is referred to the negative power supply while the operational amplifier 27 output is referred to ground, the bias current, according the present invention, consists of a D.C. component independent of the input and a low-frequency component. This component is used by the non-linear, approximately multiplicative, character of the operational transconductance amplifier 22 to operate upon the input signal.

Resistor 23 is used to lower the noise in the output of this circuit. However, it also shows an equivalency that is important to this disclosure. The net signal that drives the components 24-28 has an input component which is formed from the D.C. bias via resistor 30 and plus the input signal via resistor 23. In fact, if the transconductance amplifier 22 were a four-quadrant multiplier, instead of an approximate two-quadrant multiplier, the bias in resistor 30 could be completely replaced by the signal in resistor 23.

The current in capacitor 28 is amplified by the combination of the resistor 31 and operational amplifier 32. This amplifier produces an under damped high-pass output as the speaker does. The treble roll-off of the speaker is simulated by a low-pass filter 33.

The character of the speaker emulator can be further enhanced with a broad spectrum, low-frequency oscillator or random waveform generator 35, such as a saw-tooth, which drives the transconductance amplifier 22 via resistor 36.

THE AMPLIFIER EMBODIMENT

Figure 4 is an amplifier embodiment. The behavior of a standard tube amplifier consisting of a differential amplifier acting as a phase splitter, a pair of push-pull output tubes that drive the output via a transformer, feedback, and a power supply having main, screen, and control grid outputs is emulated by this circuit. The basic input circuit is simply copied with input coupling capacitor 41 and grid bias resistors 42 and 43. The resistor 44 corresponds to the cathode resistor of said differential phase splitter. The feedback is applied through capacitor 45 and voltage divider resistors 46 and 47. Amplifier 50 is a unity gain connected operational amplifier.

Diode 51 keeps the common cathode junction between resistors 42 and 43 from going too low. This condition occurs when the tube differential phase splitter is cutoff. Diode 52 emulates the grid conduction. Resistor 53 emulates the effective grid impedance. It may be estimated as the gain of tube phase splitter times its cathode resistance. It is adjusted to provide the desired overdrive bias shifting and resulting harmonic generation. Unity gain buffers 50 and 54 prepare the resulting signal for the generally lower impedance transconductance amplifier 55, again a Harris CA3080 or National LM3080 for example. The transconductance amplifier 55 drives inverting power amplifier 56 with a bipolar current. This amplifier has a non-linear feedback 58 to emulate the curvature of the plate resistance character. The output current, the speaker 78 current, is measured by resistor 60 and differential amplifier 61. This amplifier supplies a signal indicative of the output load current to the transconductance amplifier biasing components 62 - 67 and 71 - 76. Resistor 59 provides current feedback to amplifier 56 to give amplifier 56 a high output impedance.

Unlike the previous embodiment, the bias of the transconductance amplifier is referred to a voltage near ground by transistor 80. The primary source of the bias current and the improvement of the present invention is the current flowing in resistor 75 from the bias supply 65. The bias supply is a typical line-operated unregulated power supply. However, like the tube amplifier grid bias supply, this supply is preferably separate so that the ripple is less dependant upon the amplifier load signal.

Further, in the case that the amplifier incorporated switching supplies or was battery operated, the bias supply 65 could be replaced by an alternative broad spectrum, low-frequency source, for example, a 50 to 200 Hertz saw-tooth oscillator or a band-limited random noise generator. Note that either means is applicable to any amplifier, including a vacuum tube amplifier, that does not have the proper ripple.

The absolute value circuit 62 emulates the power supply current to the push-pull output tubes of the emulated tube amplifier.

The filter 63 emulates the response of the power supply and produces a negative going output for an increasing magnitude of output current. The character of this filter may be resonant with a frequency of about 8 hertz and a Q of about 2 or may be a low-pass filter with a time constant of approximately 100 milliseconds.

The filter 64 emulates the power tube self bias and also produces a negative going output for an increasing magnitude of output current. The character of this filter is single pole with a time constant of approximately 5 milliseconds.

The remaining path consists of another improvement of the present invention. The compressor 66 which preferably uses a series lamp, drives a low-pass filter 67. Resistor 76 passes the resulting signal to the bias input of amplifier 55.

The resistors 71 through 76 carry bias currents from the components 61 through 67 to the bias input of transconductance amplifier 55. The total bias sets the transconductance and the maximum magnitude of the output current of said transconductance amplifier. The current through resistor 71 creates even harmonics in the output because the gain is a function of the signal. The current through resistor 72 changes the gain of transconductance amplifier with the magnitude of the signal and creates odd harmonics in the output. This resistor needs to be sized to produce harmonic levels less than 1 percent at low levels and levels greater than one percent at high, but unclipped, levels. The current through resistor 73 creates the screen grid compression effect because the gain is a function of the emulated power supply response. The current through resistor 74 creates the cathode bias effects because the gain is a function of the emulated cathode bias.

Since the total current flow through resistors 71 through 76 determines the maximum current that can flow out of the transconductance amplifier 55 and drive the following amplifier 56, these should be picked with so that low impedance loads do not saturate amplifier 56 and higher impedance loads do saturate amplifier 56. This gives the amplifier its two clipping regions and a portion of the vintage tone created by worn tubes. Higher drive levels create the tone of newer tubes.

Further, resistor 72 must be sized to produce the substantial third harmonic found in push-pull amplifiers. This is sized to produce a blending of non-clipped and clipped distortion so that the amplifier distorts over a wide range of inputs. This is the opposite of the usual engineering philosophy of pushing the distortion region up to the clipping point and then paying the price of instant and harsh audible complaints.

Obviously, filter 64 and resistor 74 may be omitted if cathode or self bias effects are not wanted. However, they do produce a pleasant chime effect. Also additional filters can be added to include, for example, the output tube bias signal effects.

The absolute value circuit 62 need not be precision. The requisite diodes may exhibit their voltage drops since the effects that this circuit drives and creates occur at large signal levels. This creates an essentially linear region which then becomes non-linear as the signals approach clipping and produces the other two regions of amplifier operation.

THE RIPPLE MODULATION UPON CLIPPING

The amplifier of Figure 4A is substantially the same as Figure 4 where the numbers are common. A power supply 380-396 has been added and block 57 has been replaced with additional circuitry. The power supply begins with a secondary winding 380 and full-wave diodes 381 and 382. These diodes drive a filter capacitor 383 and attenuating resistors 384 and 385 to produce a positive voltage A that drives the base of transistor 352. In lieu of using an amplifier to produce the required inverted signal B, diodes 391 and 392 rectify the power from secondary 380 for filter capacitor 393 and attenuating resistors 384 and 385 to produce a negative voltage B that drives the base of transistor 353. Preferably these voltages are plus and minus one volt respectively with 15 percent ripple. The amount of ripple must be quite high because the it must be viewed with respect to the nominal D.C. voltage plus two diode drops and that percentage should be that of a loaded power supply in a tube amplifier or more.

The adjustable resistor 396 and the capacitor 386 adjust the level of the ripple provided to the clipping network transistors 351 and 352 without altering the average clipping level. The distortion with 396 adjusted to minimum resistance is clear while the distortion with 396 adjusted to a maximum is thick.

The network 351-357 is a non-linear network for clipping the output at a level of approximately 2.2 volts divided by the attenuation created by resistors 356 and 357. Preferably 357 is adjustable to provide a variable output. Transistors 354 and 355 buffer the attenuated signal while 353 adjusts the clipping gain and with the current feedback via 59 sets the saturation or clipping output resistance. Transistors 351 and 352 with the voltages A and B and with transistors 354 and 355 create the clipping level which includes the ripple. Since when the amplifier is clipping in the positive polarity, the ripple on voltage A helps set the clipping level. Conversely when the amplifier is clipping in the negative polarity, the ripple on voltage B helps set the clipping level. Since opposite polarities use opposite polarities of the ripple signal, this is a modulator which includes sidebands when this amplifier is clipping.

Notice that this behavior of Figure 4A occurs naturally when the output amplifier 56 clips providing the amplifier is operating at full power. This amplifier allows clipping at lower levels with the attendant ripple modulation as well.

Notice too, that the clipping circuitry with the amplifier 56 is a non-linear means having an input for the signal from amplifier 55 and the ripple signal from the power supply 380-395. The output of the amplifier includes, when clipping, the intermodulation products of these signals and consequently falls into the general description of Figure 1.

Notice that the power supply 380-395 need not be the source of the ripple, one could also use an oscillator, preferably one with a sawtooth or triangle waveform.

THE SPEAKER

The speaker 78 of Figure 4 and 218 of Figure 12 plus the speaker load for Figure 13 are preferably the fat enhanced speaker of Figure 2 which provides the desired intermodulation distortion enhancement whether the amplifier is clipping or not. Although the teachings herein suggest the use of one of the prior art speakers, such as Bussard, Lokkesmoe, or Miessner, they are not good choices and would otherwise not be used. These speakers are not in production and have not been for about 45 years, they require components that are not commonly produced, they cost more than standard speakers, and they cost more to use. Bussard and Miessner require power supply connections and larger power supplies. Bussard uses the amplifier to cancel the unwanted hum, but unfortunately, the amplifier will fail to cancel the hum when it is clipping. The Lokkesmoe speaker require substantial extra power to produce a wider bandwidth which is not needed any more. The Miessner speaker requires a third coil and the adjustment thereof to null the field coil hum.

THE POWER SUPPLY

Figure 5 is the schematic of a power supply for use in elements 35, 65, or 241 showing a center-tapped power transformer winding 91 driving two diodes 92 and 93 and filter capacitor 94 in the standard full-wave center-tapped circuit. The capacitor is sized to provide the desired embellishment, about 1 to 5 percent ripple. Figure 5A is a similar power supply but has a variable resistance load 96 for producing a variable voltage output. Optionally, the output can be buffered by an amplifier.

THE CONTROLLED GENERATOR IMPROVEMENT

Figures 3, 4, 4A and 12 may include a generator means, 35, 65, 65, and 241 respectively. Figure 6 provides an example of a generator means which may also be controlled via an input. The generator means is preferably one of the many saw-tooth oscillators known in the arts. The one illustrated in Figure 6 uses an operational amplifier 100 with both positive and negative feedback. Resistors 101 and 102 provides positive feedback and sets the voltage extremes for the saw-tooth waveform via the positive and negative saturation voltages of amplifier 100 and the attenuation of resistors 101 and 102. The negative feedback is provided by an asymmetrical RC low-pass filter 103-106. The diode 104 provides a low impedance path shunting the larger resistor 103 when the operational amplifier 100 output is low. This provides a fast discharge path for capacitor 105 via a small resistor 105. When the output of operational amplifier 100 is high, the capacitor 106 charges via resistors 105 and 103.

The output of the saw-tooth oscillator is taken from capacitor 106 either directly or via the buffer connected operational amplifier 107.

The capacitor also provides a control input 108 which is responsive to the rectifier means 111-114. A positive signal on resistor 111 drives transistor 112 in a grounded base configuration while a negative signal on resistor 111 drives the base transistor 114 with a gain limited by resistor 113. Preferably the two resistors 111 and 113 are the same value.

The low-pass filter is formed by series capacitor 121 and shunt resistor 122. This roll-off frequency of this filter should be in the higher end of the spectrum of the input to said capacitor so that the output becomes smaller with lower frequencies.

The clipping means herein is intended to clip the signal at levels below the power supply levels and should not be confused with clipping diodes often connected between the output and the power supplies to protect the amplifier against excessive output voltages.

THE RANDOM NOISE IMPROVEMENTS

Figure 7, another improvement to figures 3, 4, 4A, or 12, shows the random noise generator may be created by semiconductor noise or may be created by a pseudo-random noise generator. The pseudo-random noise generator is more consistent and more expensive than semiconductor noise. Resistor 131 is connected to a positive power supply sufficient to reverse bias the base-emitter junction of transistor 130. This will produce noise on the emitter which is amplified by circuit 132-138. The noise is capacitively coupled by capacitor 132 and the operational amplifier 134 is biased by resistor 133. The network 135-138 provides negative feedback and limits the bandwidth of the noise.

The above network could be replaced by the pseudo-random noise generator. Further, either could provide the low-frequency means output on path 4. However for greater effect the following circuit provides said output.

The capacitor 140 accepts the input to the filter control, ie. either path 6 or 7. Resistors 141 and 142 form an optional attenuator while resistor 142 biases the rectification circuit 143-146. As explained above, this circuit produces a current 147 approximately proportional to the absolute value of the input signal. This current and a minimum bias current from resistor 148 controls the bandwidth of the filter 150-151.

The controlled filter 150-151 consists of an operational transconductance amplifier 150 and capacitor 151. This amplifier. preferably a Harris CA3080 or National LM3080, is connected as a unity gain buffer. However, its transconductance is about 20 times the bias current 147. Consequently, over a range of 50 or so millivolts this amplifier appears to be a variable resistor of 0.05 volts divided by the bias current. At 100 microamps, for example, it is about 500 ohms; and at 10 microamps, 5 kilohms. The capacitor, about 1 microfarad, is picked for the desired bandwidth.

Operational amplifier 152 and feedback network 153-154 buffer and amplify the filtered signal to provide the output of the low-frequency means.

NON-LINEAR NETWORK DETAILS

The resistor-diode network of Fig. 8 is described in U.S. Patent 5,133,014. It is a plurality of parallel resistors 161-165 and series diodes 166-169. For input voltages across terminals A and B of less than one diode drop only resistor 161 conducts. For input voltages between one and two diode drops, resistors 161 and 162 conduct. Higher voltages make more resistors conduct, thereby lowering the dynamic resistance of the network.

The resistor-diode network of Fig. 9 has a plurality of parallel resistor and diode pairs in series. As the current flowing from terminal A to B increases, the voltage across the resistors increases. When the resistor voltage approaches the diode drop, the diode conducts and dynamically removes the resistor from the series string. When all of the diodes conduct, the resistance of the network is the resistance of resistor 175.

There is a rough equivalency between these networks: Equal resistors in Fig. 8 produces a current approximately proportional to the square of the voltage across the terminals. Similarly, if the resistors of Fig. 9 are in the ratios of 1, 1/2, 1/6, 1/10, 1/15...and the last resistor, the nth, is 2/n, then it too produces a current approximately proportional to the square of the voltage across the terminals A and B.

It should be noted that the networks approximate the desired function over a region. The diodes tend to sectionalize the function and eventually all of the diodes are on and the network becomes linear.

Fig. 10 also produces a squared current using semiconductor behavior found in logarithm amplifiers. The voltage across the terminals A and B is converted to a current by resistor 181. The current produces a voltage on the base of transistor 184 proportional to twice the logarithm of the current by diodes 182 and 183. The transistor 284 converts that voltage to a current in an exponential manner proportional to the square of the voltage across terminals A and B. This is made possible by biasing diode 185 with current source or large capacitor 186.

This non-linear circuit uses an active semiconductor, namely a transistor, to replace many more passive semiconductors, diodes.

For brevity in the drawings, a new symbol shown in Fig. 11 will indicate a non-linear network.

DESCRIPTION OF ANOTHER AMPLIFIER EMBODIMENT

The tube amplifier behavior is provided by the circuit shown in Fig. 12. It shows a complementary "phase splitter" and bipolar push-pull output which emulates push-pull pentodes with a poorly regulated power supply. Fig. 12 is a combination of Figures 10 and 11 of a preceding application, now U.S. patent 5,434,536.

The components 191 through 199 is an approximation to the phase splitter for a bipolar amplifier which requires both inputs in-phase. Since the two triodes in a differential amplifier phase splitter compensate each other, the stage produces very little distortion until clipping. The output resistance of the phase splitter is about twice the triode plate resistance normally, but becomes nearly infinite when clipping.

When the output of amplifier 191 goes high, network 192 pulls up voltage at 193. When the voltage at 193 approaches the plate voltage P, network 194 becomes more resistive and disconnects when the voltage at 193 is greater than the voltage at P. At the same time, network 195 disconnects and the current from source 196 flows through network 197 to plate N. Symmetric behavior occurs when the amplifier 191 output goes negative: network 192 disconnects, P has current from current source 198, network 195 pulls down voltage at 199, and network 197 disconnects from plate N. The networks 192, 194, 195, and 197 use an extra diode in series with the input to keep reverse currents from flowing.

The components 191-199 of Fig. 12 provides the soft cutoff for the grid circuit of the output stage. Since the negative half of the output stage operates symmetrically to the positive half, only the positive (upper) half will be detailed. As shown, the lower half operates in phase with the non-linearities in the opposite direction. Resistor 202 is the plate resistor for the input circuit. Capacitor 203 is the coupling capacitor that connects the plate terminal P to the grid terminal G of the following tube emulator. Diodes 204 and 205, connected to the grid terminal, emulate positive grid conduction. Zener diode 205 adjusts for the nominal zero bias of this stage, and in general represents a voltage offset source such as found in Figure 5A. Resistor 206 is the grid resistor which drives amplifier 208 with feedback resistor 207. Network 209 is nominally a squaring, second order emulation of the pentode transfer characteristic. This gain varying characteristic provides smooth crossover and the variable gain for emulating tube compression. Amplifier 211, shown as a transistor, shifts the level of the signal to the output supply voltage +40 with the help of resistor 212. MOSFET 213 with source resistor transfers the voltage on resistor 212 to a current through resistor 214. Bias resistor

210 is adjusted to overcome the threshold voltage of MOSFET 213. The remaining bias is established by the voltage on the base of transistor 211. Zener diode allows the load to fly back some before it is clamped.

The components 203-206 form a bias shifter. The diodes correspond to the grid conduction of tubes. The capacitor 203 corresponds to the coupling capacitor. And resistor 206 corresponds to the grid resistor.

Inverting amplifier 220 and non-linear networks 221 and 222 feedback the output to emulate the plate resistance of a pentode. Notice that the feedback loop goes through both non-linear networks. Consequently, the plate resistance and the transfer characteristics are functions of both the output and the input. This is seen in the different slopes of pentode plate curves.

The saturation region is emulated by resistor 214. Again, the entire characteristic is not perfect, but around the load line it is a good approximation.

The poor regulation of the power supply coupled with screen grid operation creates the compression found in tube amplifiers. When the power supply sages under the load of large signals, the screen voltage goes down in a manner dictated by the power supply filter. The drop in screen voltage lowers the output current and lowers the gain of the tube.

The screen grid voltage shift can be lumped into a control grid shift according to Thomas Martin in his book *Electronic Circuits*, Prentice-Hall, pages 84-87 providing the signal is scaled appropriately.

Although the power supply could drive this circuit, it is simpler to estimate the power current with filter 230. The resulting signal is rectified by 231 and then filtered by 232 which has the same time constants and overshoot as the emulated power supply. The output of 232 is fed to the negative half by resistor 235 while being inverted by 233 and fed to the positive half by resistor 234. An increasing output then reduces the bias on networks 209 and 236 reduces the output currents, increases the resistance of these networks and lowers the gain. The compression control signal from the output of filter 232 is canceled in the output.

The use of the power supply 5A in lieu of the zener diode 205 provides two features. First, the ripple of the power supply creates an intermodulation when the diode 204 is conducting. Second, the variable resistor 96 can vary the clipping level and consequently the output power. Of course, the opposite zener is then replaced with the opposite polarity supply.

The speaker 218 is preferably the fat enhanced speaker of Figure 2 which provides the desired intermodulation distortion whether the amplifier is clipping or not.

THE VARIABLE RESISTANCE EMBODIMENT

There are several candidates for the variable resistance embodiment, the light dependent resistor, the field effect transistor, and a non-linear device or network. The light dependent resistor is used by the sub-sonic tremolo circuit, and would work in the present invention if capable of the speed. The field effect transistor is known for its variable resistance region, its only problem with the field effect transistor is its production variability. However, adequate selection can produce a suitable non-linear means.

Figure 13 shows a non-linear device embodiment that uses the variability of the dynamic resistance to create intermodulation. Amplifier 250 and non-linear network 251 is representative of a triode tube emulator driving stage having plate resistor 252. The signal on the plate is coupled by capacitor 253 and biased by resistor 254 to the phase splitter transistor 255. Transistor 255 is representative of a cathodyne phase splitter or its emulator and has load resistors 256 and 257. The two phases are coupled with capacitors 260 and 261 to the grid terminals G of two triode emulators. The grid conduction is created by diodes 262 and 263 connected to the grid terminals and conduction offset device 264, a zener diode for example. The grid resistors 266 and 267 are also the input resistors for inverting amplifiers 268 and 269 that have gain setting feedback resistors 270 and 271. The inverting amplifiers are coupled to an output transformer 274 by non-linear networks 272 and 273. The center tap of the output transformer is powered by a relatively poor power supply 275, such as Figure 5 that has a desirable amount of ripple. Alternatively, one could also use the circuits of Figures 6 or 7 providing they were properly biased at approximately half the positive supply voltage.

The ripple signal from source 275 interacts with non-linear networks 272 and 273 to produce opposing currents in the transformer which cancel to a degree determined by the equivalence of the resistance of the non-linear networks 272 and 273. When the input signal causes the networks to have unequal resistances, the ripple signals in the transformer are not equal and consequently produces an output. Since the resistance difference and consequently the ripple signal output is determined by the signal, the ripple signal output is the intermodulation products of the input and the ripple signal.

Optionally, the conduction offset device 264 can be the power supply of Figure 5A. The ripple in this supply produces intermodulation when the diodes 262 and 263 conduct. The variable resistance controls the clipping level of the grid signals on terminals G. Notice that this technique works for tube amplifiers as well providing the polarity of the Figure 5A power supply is negative.

This approach does not work in typical solid state for many reasons: 1) the typical transistor has a very high output resistance, 2) the output resistance is made higher with emitter resistors, 3) the typical output stage is an emitter or drain follower, 4) the amplifier uses substantial feedback, or 5) the power supply has very little ripple. Tube amplifiers, particularly triodes, have a lower output resistance, typically do not have cathode degeneration, typically are not cathode followers, do not use nearly as much feedback, and have sizeable ripple in the power supplies. By experimentation, this and/or screen grid influences produce the tube embellishment.

Figure 13 also shows a second input to amplifiers 268 and 269 via resistors 280 and 281 from a limited upper spectrum source or filter 282. While 275 alters the transformer side voltages of the non-linear networks, 282 alters the amplifier side voltages. The results are the same until the amplifiers clip where the variation from 275 continues while the embellishments from 282 are periodically interrupted by the clipping behavior.

Notice that this form is the same required to enhance any non-linear push-pull amplifier, including tube amplifiers. Notice further, that this is quite similar to the network 241-245 used to enhance a complementary, non-linear push-pull amplifier.

Just as the injection of a ripple signal produces a clipping intermodulation in the circuit of Figure 4A, the inclusion of a ripple signal in the conduction offset device 264 also produces clipping intermodulation.

Adjusting the offset source voltage downward reduces the output signal because diodes 262 and 263 limit the output signal. This structure is applicable to all amplifiers including vacuum tube amplifiers.

THE COMPUTER EMBODIMENT

Figures 14 and 15 address the ever growing digitalization of the world including audio. Figure 14 shows an analog-to-digital converter 191 providing digital signals to a computer 192. The computer provides digital signals to a digital-to-analog converter 193. The input is sampled periodically, converted to digital, operated upon by the computer, and converted back to digital. Since the same program is executed by the computer for each sample, it is only necessary to indicate the processing for a single sample.

The well-known arts for storing and transmitting digital data may remove the converters from direct connection to said computer.

Figure 15 shows a flow chart for the single sample programming. The program starts at 194, computes the low-frequency signal in 195, performs the non-linear mathematics in 196, and returns program control in 197.

The low frequency signal may be computed in 195 by techniques within the digital arts. It may be a filtered version of the input or the output or is a digitally created signal. A digitally created signal, particularly a saw-tooth is simply created by incrementing a value V with a value INC at each sample time in Fortran:

```
V = V + INC
```

The natural overflow will make V appear as the desired saw-tooth. At this point the variable V may be as shown, the input value, or the output value. However, a saw-tooth has an infinite spectrum and the input or the output has too great a spectrum. Any of these needs to be limited with a filter. There are many digital filters. For simplicity, this is an infinite response type that uses the value F for a filter constant. LFF is the output of the filter and LFS is the value for the low-frequency source:

LFF = LFF + F * (V - LFF)
LFS = LFF + BIAS

This saw-tooth can be controlled in frequency by making INC a variable dependent upon the absolute value of the output of a digital filter. The digital filter responds to either the input or the output.

The programming for step 196 is also quite simple:

OUTPUT = INPUT * LFS

Please note that mathematics is often distributive and this value is equivalent to which unfortunately has a quite different description. This situation is like the effect of resistor 23 in Figure 3.

OUTPUT = INPUT * BIAS + INPUT * LFF

This concept may be generalized to the circuitry within by applying well-known circuit analysis techniques to the figures within. This disclosure shows circuitry whose operational characteristics are well-known and readily translated to digital programming since their functions are within the digital processing arts.

LESS LOW-FREQUENCY SIGNAL

The operation of the non-linear means and the low-frequency signal means with the D.C. bias produces the input signal and intermodulation products of the input and the low-frequency signal. However, the little or no D.C. bias on the input and nearly perfect operation of a multiplier means produces little or no low-frequency signal at the output. The gain of the low-frequency is substantially smaller than the gain of the input signal, hence there is less low-frequency signal than input signal in the output.

CLAIMS:

1. An audio amplifier having a first input for receiving an input signal having a first audio spectrum having at least one audio frequency, having an output for producing an output signal, having feedback with a loop gain and a feedback signal, and having a gain between said input and output, comprising:

input means for receiving said first input and said feedback signal and for producing an intermeans signal;

power control means for controlling the maximum level of said intermeans signal and the gain of the amplifier;

output means responsive to said intermeans signal for producing said output and said feedback signal; and

feedback control means operating in conjunction with said power control means for keeping said loop gain constant.

2. The amplifier of claim 1 including clipping means for clipping said output wherein the clipping level of said clipping means remains constant with respect to said first input.

3. The amplifier of claim 1 wherein said input means includes a variable gain amplifier having a gain responsive to a gain control signal having a second audio spectrum and wherein said output has a spectrum which has said audio frequency plus sidebands created by said second audio spectrum.

4. The amplifier of claim 3 wherein said gain control signal is responsive to an amplifier signal.

5. The amplifier of claim 3 including a power supply having ripple and wherein said gain control signal is responsive to said power supply.

6. The amplifier of claim 5 including load means responsive to an amplifier signal for loading said power supply having ripple to increase said ripple as a function of said amplifier signal.

7. The amplifier of claim 1 wherein said input means is responsive to a computed signal and includes a digital means comprising:

analog-to-digital conversion means for receiving said first input and producing a first digital output;

computer means having programs for calculations upon said first digital output for producing a second digital output and for controlling said power control means and said feedback control means; and

digital-to-analog conversion means for receiving said second digital output for producing said computed signal.

8. The amplifier of claim 7 wherein said computer controls said power control and said feedback control with a servo controlled potentiometer means.

9. The amplifier of claim 7 wherein said computer controls said power control and said feedback control with light dependant resistor means.

10. The amplifier of claim 7 wherein said computer controls said power control and said feedback control with multiplexer means.

11. The amplifier of claim 1 wherein said input means includes non-linear means for producing harmonics without clipping and for emulating the cutoff of triode vacuum tubes.

12. The amplifier of claim 1 wherein said input means includes low-pass filter means for filtering said input means.

13. The amplifier of claim 1 wherein said input means includes bias shifting means for altering the bias of a signal in said input means in response to the amplitude of said signal.

14. The amplifier of claim 1 wherein said input means includes compression means.

15. The amplifier of claim 1 wherein said output has a frequency response which is reduced upon the overdrive of the input means by large signals on said input.

16. The amplifier of claim 1 wherein said output means has an output impedance which is inductive at low audio frequencies, resistive at middle audio frequencies and capacitive at high audio frequencies.

17. The amplifier of claim 1 which includes digital computer control means to control said power control and feedback control means.

18. An audio amplifier having no thermionic emission devices, having a first input, having an output and an output signal, having an audio input-to-output frequency response characteristic, and having an output impedance comprising:

an input means for receiving said first input and a feedback signal, for producing an intermeans signal, and for disabling said feedback signal when a signal on said input overdrives said input means;

an output means responsive to said intermeans signal for producing said output signal and said feedback signal;

said output means having an output impedance dependant upon the feedback signal when the feedback signal is disabled the output impedance is capacitive at high frequencies, resistive at middle frequencies, and inductive at low frequencies and when the feedback is present the output impedance is reduced and proportionally more resistive.

19. The amplifier of claim 18 wherein said input means includes non-linear means for altering said output impedance as a function of the signal on said output.

20. The amplifier of claim 18 wherein said input means includes non-linear means for altering said output impedance as a function of the signal on said input.

21. The amplifier of claim 18 wherein said input means includes a variable gain amplifier means which is responsive to one or more gain control signals.

22. The amplifier of claim 21 wherein said input means includes ripple means for creating power supply ripple and for producing a gain control signal.

23. The amplifier of claim 18 wherein said input means includes bias shifting means for altering the bias of an input means signal as a function of said signal.

24. The amplifier of claim 18 wherein said input-to-output frequency response narrows upon the disabling of the feedback by excessive signals.

25. The amplifier of claim 18 wherein said input means is responsive to a computed signal and includes:

analog-to-digital conversion means for receiving said input and producing a first digital output;

computer means having programs for calculations upon said first digital output for producing a second digital output; and

digital-to-analog conversion means for receiving said second digital output for producing said computed signal.

26. An audio amplifier having an input having a first audio spectrum having at least one audio frequency component and an audio output having an output impedance comprising:

input amplifier means responsive to said input;

first control means for controlling the output of said input amplifier;

output amplifier responsive to said input amplifier and responsive to a clipping signal; and

clipping means responsive to said output amplifier for producing said clipping signal wherein the clipping level of said clipping means is controlled by a second control means; wherein

said output impedance is inductive for low audio frequencies, resistive for middle audio frequencies, and capacitive for high audio frequencies.

27. The amplifier of claim 26 wherein said input amplifier is also responsive to feedback and said feedback is controlled by a control means.

28. The amplifier of claim 26 wherein said first and second control means operate simultaneously.

29. The amplifier of claim 26 which includes computer means for controlling said first and second control means.

30. The amplifier of claim 26 which includes:

analog-to-digital conversion means responsive to said input for producing digital representations of the input;

computer means responsive to said analog-to-digital conversion means for creating sound effects and for producing digital representations therefor; and

digital-to-analog conversion means responsive to said computer means for driving said input amplifier.

31. The amplifier of claim 26 wherein said output amplifier includes:

current feedback means for sensing the current in said output and producing a current feedback signal;

connection means for combining current feedback signals and input amplifier signals;

capacitive coupling means for coupling said input and current feedback signals to an output amplifier;

voltage feedback means responsive to said output for producing voltage feedback signals; and

output amplifier means responsive to said coupled input and current feedback signals and responsive to voltage feedback signals for producing said output.

32. The amplifier of claim 31 wherein said voltage feedback means includes low-pass filter means.

33. The amplifier of claim 31 wherein said connection means includes a shunt resistor connected to ground.

34. The amplifier of claim 31 wherein said input means includes a variable gain amplifier responsive to said input and responsive to gain control signals.

35. The amplifier of claim 31 wherein said input means includes amplifier means responsive to said input for providing gain and non-linear means connected to the output of said amplifier means for providing the cutoff characteristic of triode vacuum tubes.

36. The amplifier of claim 26 wherein said input amplifier includes a transconductance amplifier with a gain control input responsive to said one or more control means.

37. The amplifier of claim 36 which includes a power supply having ripple to produce a gain control signal.

38. The amplifier of claim 36 which includes a first filter means responsive to said output for producing a gain control signal.

39. The amplifier of claim 36 which includes a rectification means responsive to said output for the production of a gain control signal.

40. The amplifier of claim 39 which includes a second filter means responsive to said rectification means for the production of a gain control signal.

41. The amplifier of claim 36 which includes an oscillator means for the production of a gain control signal.

42. The amplifier of claim 26 wherein said input means includes a first non-linear means for producing harmonics without clipping and for emulating the cutoff of triode vacuum tubes.

43. The amplifier of claim 42 wherein said input means includes solid state means for emulating a cathodyne phase splitter.

44. The amplifier of claim 42 wherein said input means includes:

a phase splitter having first and second phase splitter outputs; and

first and second coupling capacitors for coupling said first and second phase splitter outputs to first and second diode means and to second and third non-linear means; wherein

said second and third non-linear means drive said output amplifier.

45. The amplifier of claim 26 wherein said clipping level control is a variable attenuator responsive to said output and wherein said clipping means includes an emitter follower transistor which drives a grounded base transistor.

46. An audio amplifier having an input with a first spectrum with at least one frequency component and having an output with an output impedance comprising:

an amplifier responsive to said input, for providing said output, having one or more gain control means, and having clipping means with clipping level control means; and

control means independent of said output for conjunctively controlling the gain of said amplifier and the clipping level of said amplifier so that the clipping level relative to the level of said input is approximately constant.

47. The audio amplifier of claim 46 wherein said amplifier is also responsive to a feedback signal derived from said output thereby establishing a feedback loop with a loop gain and said control means maintains said loop gain and said output impedance approximately constant.

48. The audio amplifier of claim 46 wherein said gain and clipping level control means includes computer means.

49. The audio amplifier of claim 46 wherein said amplifier is responsive to a computed signal and said audio amplifier includes:

analog-to-digital conversion means for receiving said input and producing a first digital output;

computer means having programs for calculations upon said first digital output for producing a second digital output; and

digital-to-analog conversion means for receiving said second digital output for producing said computed signal; wherein

said computer means also operates said control means.

50. The audio amplifier of claim 46 wherein said amplifier includes a source of a second audio spectrum and the gain of said amplifier is responsive to said source to produce intermodulation products of said first and second audio spectra and wherein said second spectrum gives said frequency component sidebands.

51. The audio amplifier of claim 46 wherein said amplifier has an output impedance which is inductive at low audio frequencies, resistive at middle audio frequencies, and capacitive at high audio frequencies.

52. The audio amplifier of claim 46 which includes an input amplifier for receiving said input and driving said amplifier, said input amplifier including an operational amplifier means for providing gain and a non-linear means for emulating the plate characteristics of a vacuum tube.

53. The audio amplifier of claim 46 wherein said amplifier includes:

solid state amplifier means for receiving said input and for providing gain;

a non-linear means connected to the output of said solid state amplifier means for emulating the plate characteristics of a vacuum tube; and

transistor means connected to said first non-linear means for emulating a cathodyne phase splitter.

54. An audio amplifier having an input with a first spectrum with at least one frequency component and having an output with an output impedance comprising:

an amplifier responsive to said input and a clipping signal for providing said output;

clipping means including an emitter follower transistor responsive to said output and a grounded base transistor for producing said clipping signal; and

clipping level control means for adjusting the clipping level of said amplifier output

55. The audio amplifier of claim 54 wherein said clipping level control means is a variable attenuation means connected between said output and said clipping means.

56. The audio amplifier of claim 54 wherein said amplifier includes one or more gain control means and control means for controlling the gain of said amplifier and the clipping level of said amplifier so that the clipping level relative to the level of said input is approximately constant.

57. The audio amplifier of claim 54 wherein said amplifier is also responsive to a feedback signal derived from said output thereby establishing a feedback loop with a loop gain and said control means maintains said loop gain and said output impedance approximately constant.

58. The audio amplifier of claim 54 wherein said amplifier includes a source of a second audio spectrum and the gain of said amplifier is responsive to said source to produce intermodulation products of said first and second audio spectra and wherein said second spectrum produces sidebands to said frequency component.

59. The audio amplifier of claim 54 wherein said amplifier has an output impedance which is inductive at low audio frequencies, resistive at middle audio frequencies and capacitive at high audio frequencies.

60. The amplifier of claim 54 wherein said amplifier is responsive to a computed signal and said audio amplifier includes:

analog-to-digital conversion means for receiving said input and producing a first digital output;

computer means having programs for calculations upon said first digital output for producing a second digital output; and

digital-to-analog conversion means for receiving said second digital output for producing said computed signal; wherein

said computer means also operates said control means.

61. A solid state audio amplifier having an input and an output with an output signal comprising:

input means for receiving said input and producing an input signal;

current feedback means for sensing the current in said output and producing a current feedback signal;

connection means including capacitive coupling means for connecting said input and said current feedback signals to produce a capacitively coupled combined signal;

voltage feedback means responsive to said output for producing voltage feedback signals; and

output amplifier means responsive to said coupled combined signal and responsive to voltage feedback signals for producing said output;

wherein said output has an output impedance which is inductive at low audio frequencies and resistive at medium audio frequencies.

62. The amplifier of claim 61 wherein said voltage feedback means includes low-pass filter means.

63. The amplifier of claim 61 wherein said input means includes a variable gain amplifier responsive to said input and responsive to gain control signals.

64. The amplifier of claim 61 wherein said input means includes amplifier means responsive to said input for providing gain and non-linear means connected to the output of said amplifier means for providing the cutoff characteristic of triode vacuum tubes.

65. The amplifier of claim 61 wherein said input means includes:

amplifier means responsive to said input for providing gain;

non-linear means connected to the output of said amplifier means for providing the cutoff characteristics of triode vacuum tubes; and

transistor means connected to said non-linear means for emulating a cathodyne phase splitter.

66. The amplifier of claim 61 wherein said output amplifier has an output impedance which is inductive at low audio frequencies, resistive at middle audio frequencies and capacitive at high audio frequencies.

67. An audio amplifier system improved by an intermodulation distortion enhancement means having an audio input and an audio output for producing an audio output signal enhanced by intermodulation products comprising:

a limited upper audio spectrum source means for producing upper audio spectrum signals having a decreasing upper audio spectrum which has a minus 3 db frequency above 50 Hertz and which is a rectifier-less filter means for filtering one of said audio input or said audio output wherein said filtered signal D.C. component is substantially independent of an audio portion of said audio input; and

a non-linear means responsive to said audio input and said audio spectrum source means for producing said audio output signal which includes, whether clipping or not clipping, signal components of the audio input and intermodulation products of said audio input and said limited spectrum source means and comparatively less of the signal from said audio spectrum source means.

68. The intermodulation distortion means of claim 67 wherein said filter means is responsive to said audio output.

69. The intermodulation distortion means of claim 67 wherein said filter means is responsive to said audio input.

70. The intermodulation distortion means of claim 69 including speaker means having permanent magnet means operable over the operating range of said speaker means, voice coil means connected to the output of said amplifier, a field coil means mechanically isolated from said voice coil for altering the magnetic field at said voice coil connected to said independent upper spectrum limited source means; wherein

the non-linear means is the multiplicative relationship between the current in said voice coil and the magnetic field at the voice coil produced by said permanent magnet means and said field coil means;

the filter means includes said field coil; and

the interaction of said voice coil and said permanent magnet produces a greater output than interactions with said field coil.

71. The intermodulation distortion means of any one of claims 67 - 70 wherein said audio spectrum source means which includes compression means.

72. The intermodulation distortion means of any one of claims 67 - 70 wherein said non-linear means is connected to said input by a filter means for removing bass frequencies.

73. An audio amplifier system improved by an intermodulation distortion enhancement means having an audio input and an audio output for producing an audio output signal enhanced by intermodulation products comprising:

a limited upper audio spectrum source means, for producing upper audible spectrum signals having an audible fundamental and having a decreasing upper audio spectrum which has a minus 3 db frequency above 50 Hertz and which is present whether or not said audio input is present; and

a non-thermionic emission, non-linear means means responsive to said audio input and said audio spectrum source means for producing said audio output signal which includes signal components of the input and intermodulation products of said audio input and said audio spectrum source means and comparatively less of the signal from said audio spectrum source means.

74. The intermodulation distortion means of claim 73 wherein said non-linear means is an amplifier and a clipping means for clipping the output of said amplifier.

75. The intermodulation distortion means of claim 74 wherein said clipping means is connected to said amplifier by a variable attenuator.

76. The intermodulation distortion means of claim 74 wherein said intermodulation products are produced whether clipping or not clipping.

77. The intermodulation distortion means of claim 76 wherein said audio spectrum source means includes a power supply means for providing at least a portion of said upper audio spectrum signal.

78. The intermodulation distortion means of claim 76 wherein said audio spectrum source means includes a variable power supply means for providing at least a portion of said upper audio spectrum signal.

79. The intermodulation distortion means of claim 76 wherein said audio spectrum source means includes an oscillator means.

80. The intermodulation distortion means of claim 76 wherein said audio spectrum source means includes an oscillator means which produces a saw-tooth or triangle waveform.

81. The intermodulation distortion means of claim 76 wherein said audio spectrum source means includes an oscillator means that produces a frequency, in the 100 to 300 Hertz range.

82. The intermodulation distortion means of claim 76 wherein said audio spectrum source means includes a random noise generator means.

83. The intermodulation distortion means of any one of claims 67 - 69 or 76 - 82 wherein said non-linear means a variable resistance means controlled by a signal means.

84. The intermodulation distortion means of any one of claims 67 - 69 or 76 - 82 for the emulation of speakers having non-linear means responsive to said audio spectrum source means and which includes a low-pass filter for emulating the treble roll-off of the emulated speaker.

85. The intermodulation distortion means of any one of claims 67 - 69 or 76 - 82 for the emulation of speakers and having non-linear means responsive to said audio spectrum source means and including high pass filter means.

86. An audio amplifier system improved by an intermodulation distortion enhancement means having an audio input and an audio output for producing an audio output signal enhanced by intermodulation products comprising:

a limited upper audio spectrum source means which is a full-wave power supply; and

a push-pull non-linear means responsive to said audio input and said audio spectrum source means for producing said audio output signal which includes, signal components of the audio input and intermodulation products of said audio input and said audio spectrum source means and comparatively less of the signal from said audio spectrum source
means; wherein

each of said push-pull non-linear means has an input responsive to both said audio input and said audio spectrum source means and said power supply provides with at least 1 percent ripple as measured at the inputs to said non-linear means, and said amplifier means includes an amplifier means and non-linear means for the emulation of a triode vacuum tube.

87. An audio amplifier system improved by an intermodulation distortion enhancement means having an audio input and an audio output for producing an audio output signal enhanced by intermodulation products comprising:

a limited upper audio spectrum source means which is a full-wave power supply; and

a push-pull non-linear means responsive to said audio input and said audio spectrum source means for producing said audio output signal which includes, signal components of the audio input and intermodulation products of said audio input and said audio spectrum source means and comparatively less of the signal from said audio spectrum source
means; wherein

each of said push-pull non-linear means has an input responsive to both said audio input and said audio spectrum source means and said power supply provides with at least 1 percent ripple as measured at the inputs to said non-linear means, and said amplifier means includes an amplifier means and non-linear means for the emulation of a pentode vacuum tube.

88. An audio amplifier system improved by an intermodulation distortion enhancement means having an audio input and an audio output for producing an audio output signal enhanced by intermodulation products comprising:

a limited upper audio spectrum source means which is a full-wave power supply; and

a push-pull non-linear means responsive to said audio input and said audio spectrum source means for producing said audio output signal which includes, signal components of the audio input and intermodulation products of said audio input and said audio spectrum source means and comparatively less of the signal from said audio spectrum source
means; wherein

each of said push-pull non-linear means has an input responsive to both said audio input and said audio spectrum source means and said power supply provides with at least 1 percent ripple as measured at the inputs to said non-linear means, and said amplifier means includes means for emulating a vacuum tube amplifier power supply under load.

89. The intermodulation distortion means of any one of claims 86 - 88 wherein said power supply is filtered with no more than one capacitor per power supply output.

90. The intermodulation distortion means of any one of claims 67 - 69 or 73 - 82 wherein said amplifier means includes an amplifier means and non-linear means for the emulation of a triode vacuum tube.

91. The intermodulation distortion means of any one of claims 67 - 69 or 73 - 82 wherein said amplifier means includes an amplifier means and non-linear means for the emulation of a pentode vacuum tube.

92. The intermodulation distortion means of any one of claims 67 - 69 or 73 - 82 wherein said amplifier means includes means for emulating a vacuum tube amplifier power supply under load.

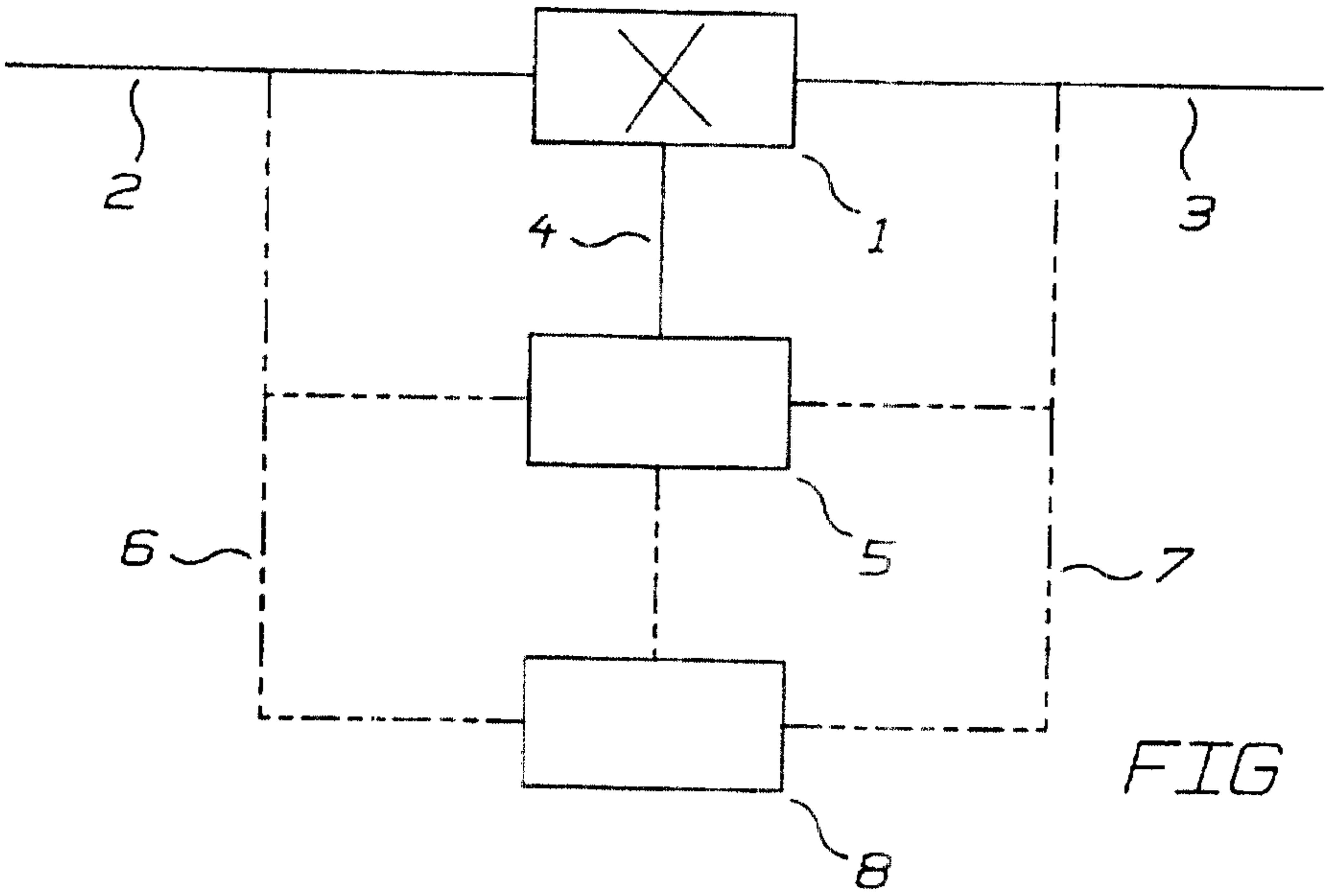

FIG 1

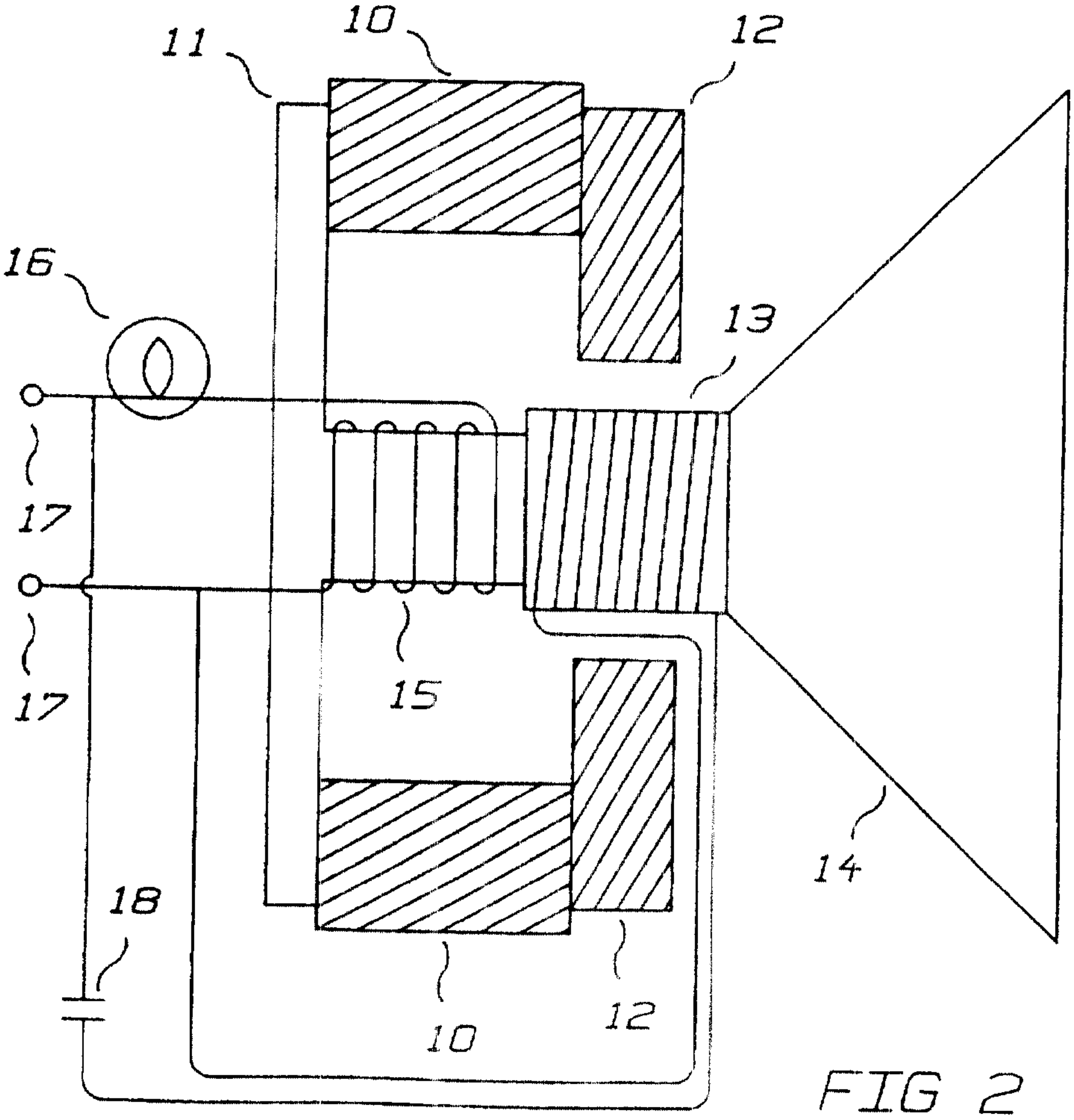

FIG 2

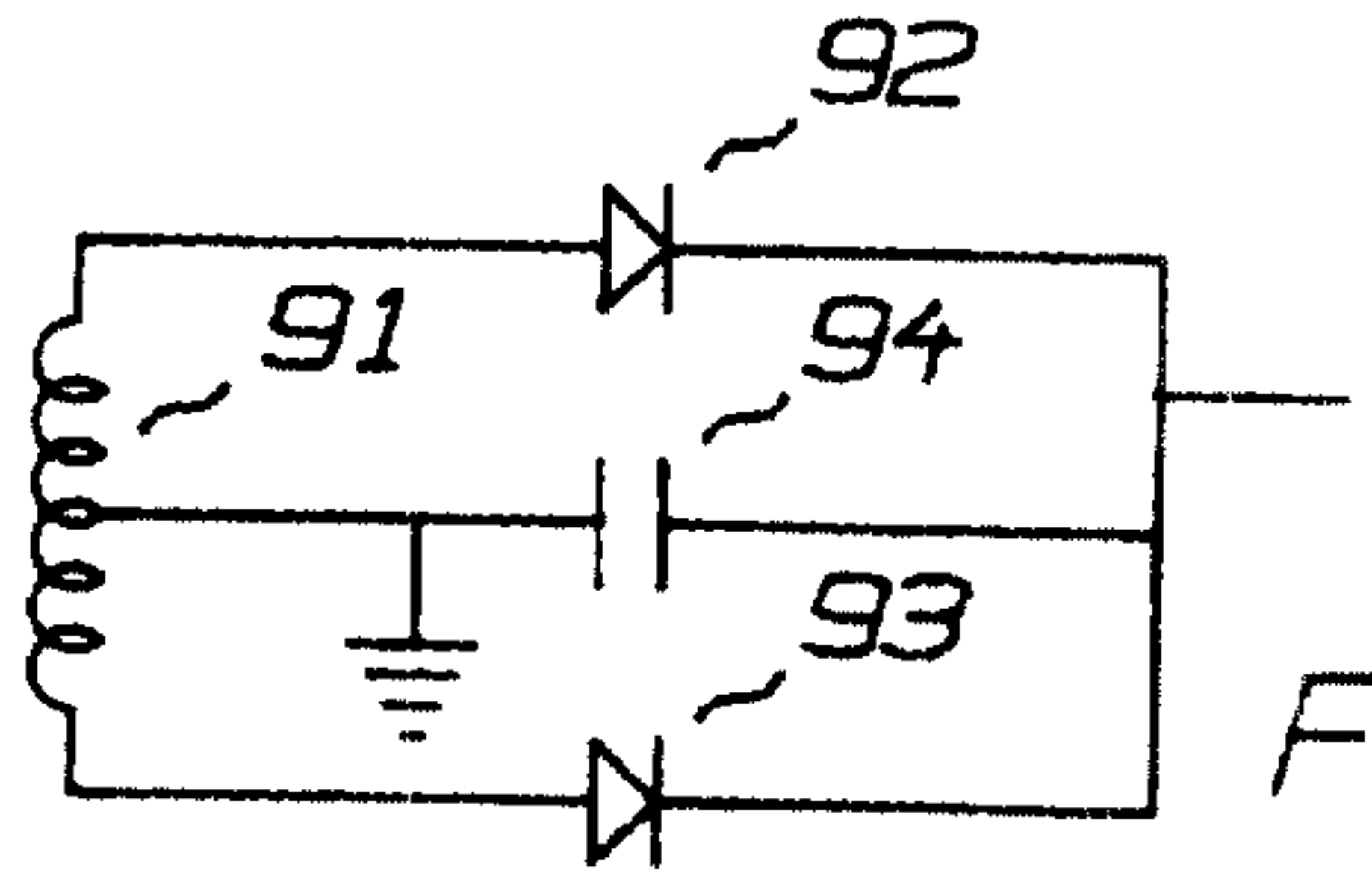

FIG 5

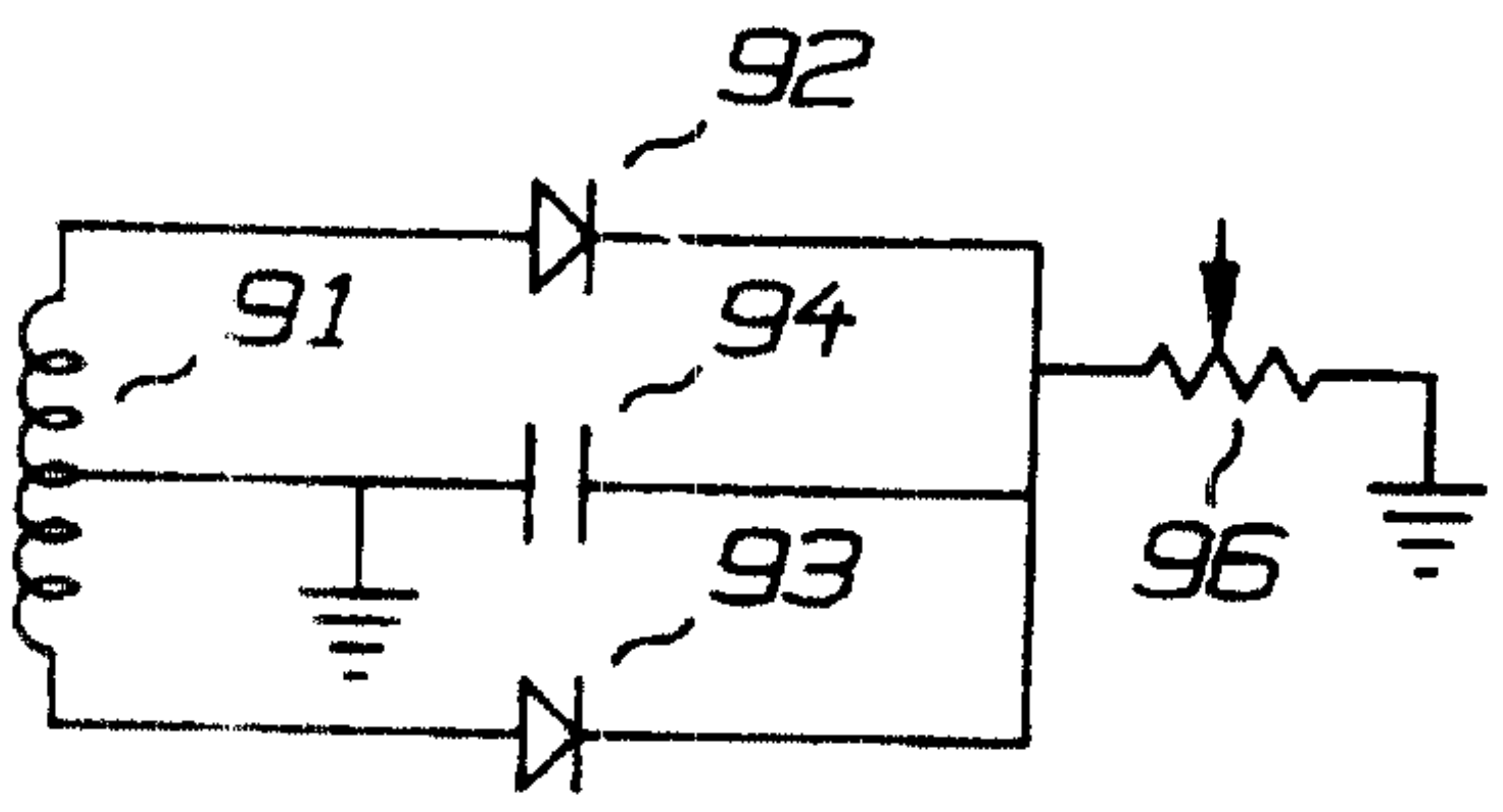

FIG 5A

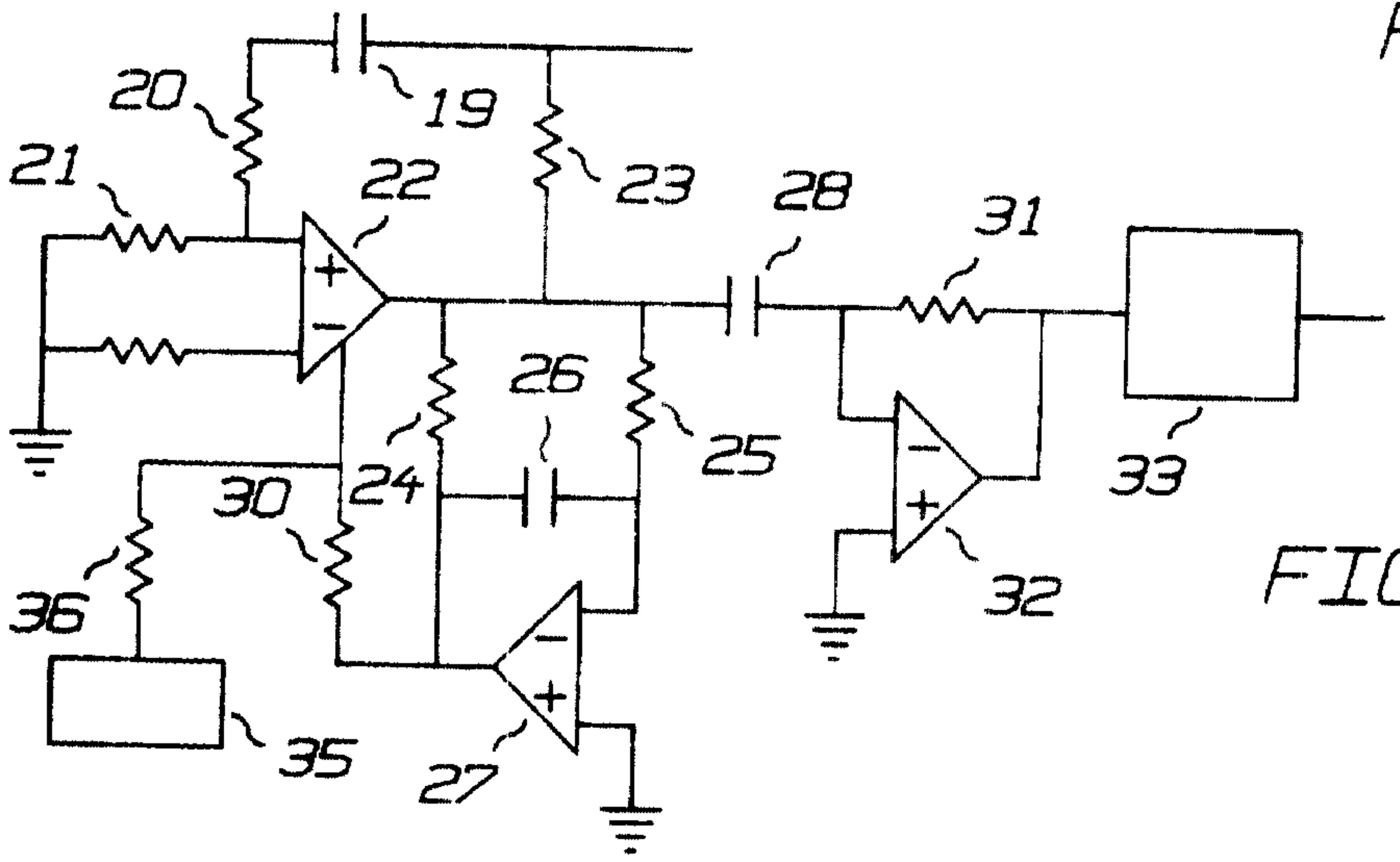

FIG 3

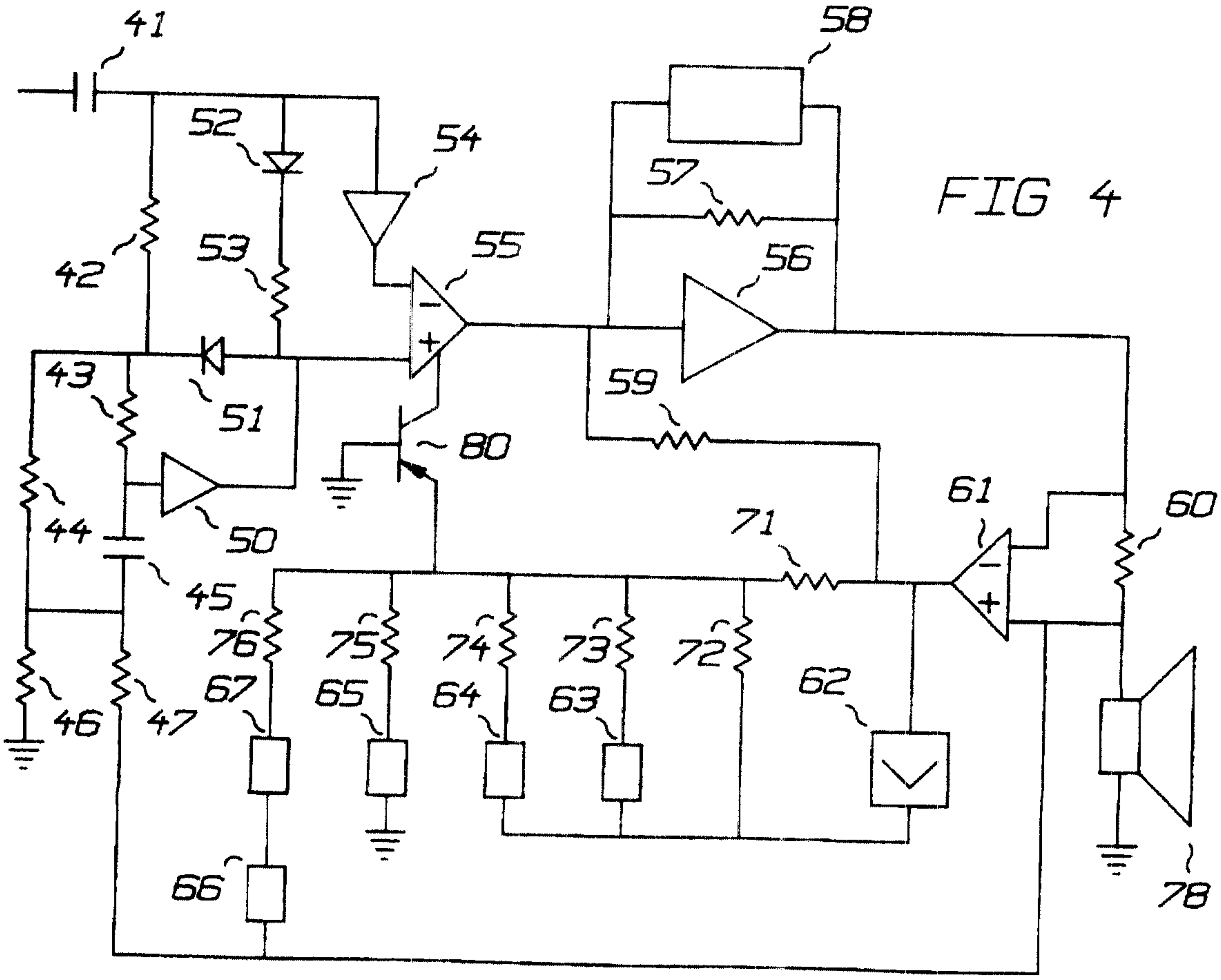

FIG 4

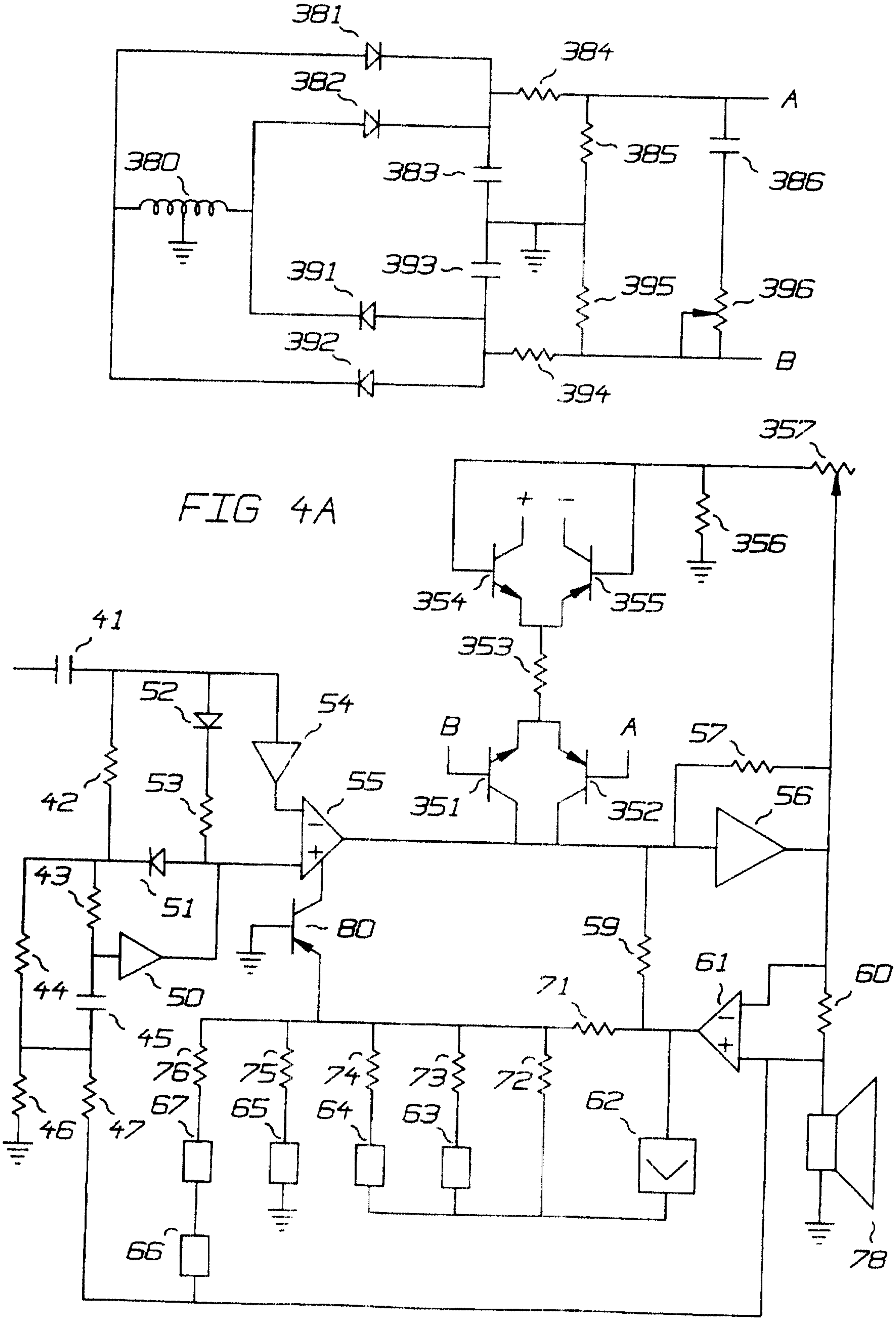
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FIG 4A

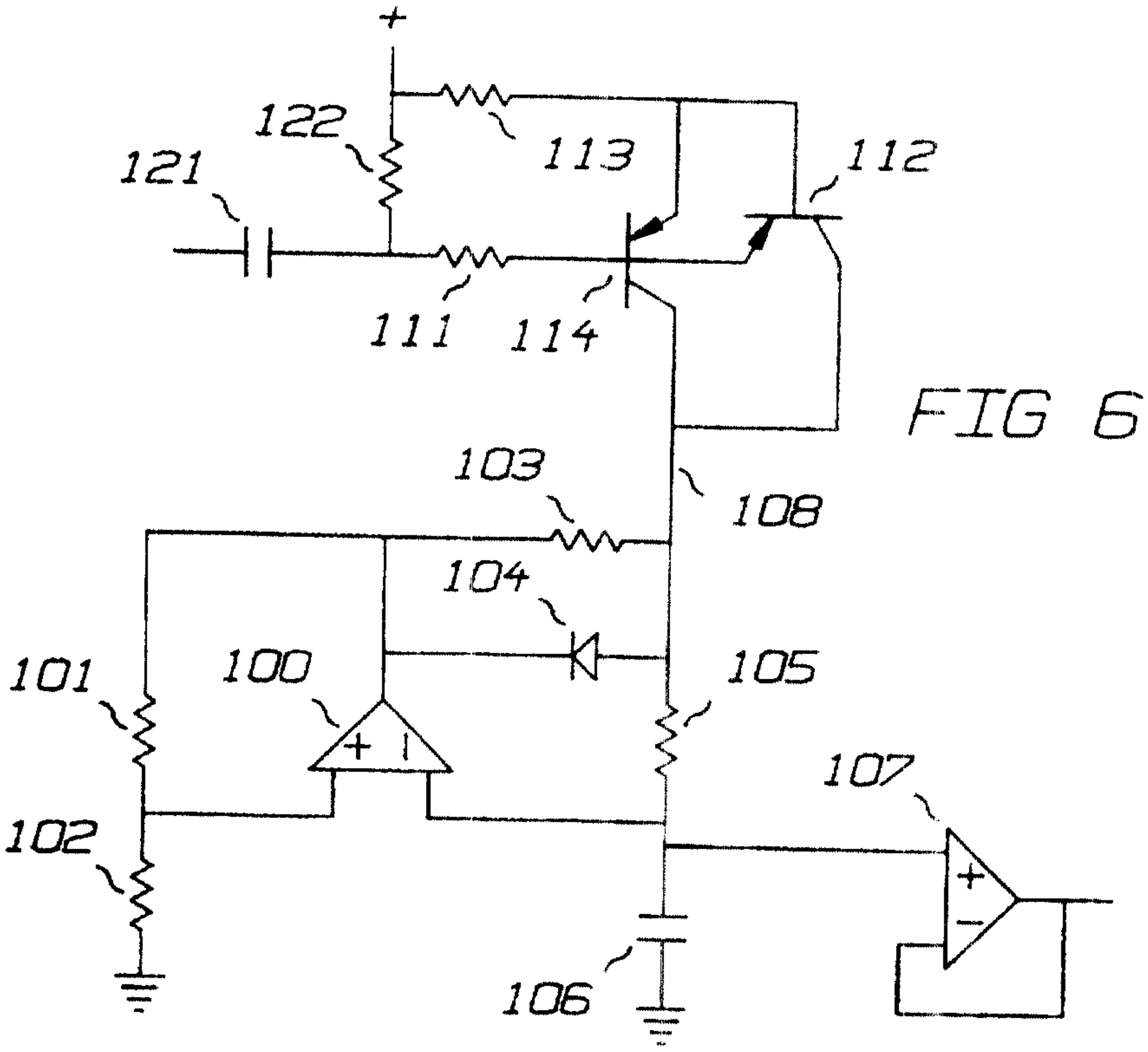
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FIG 6

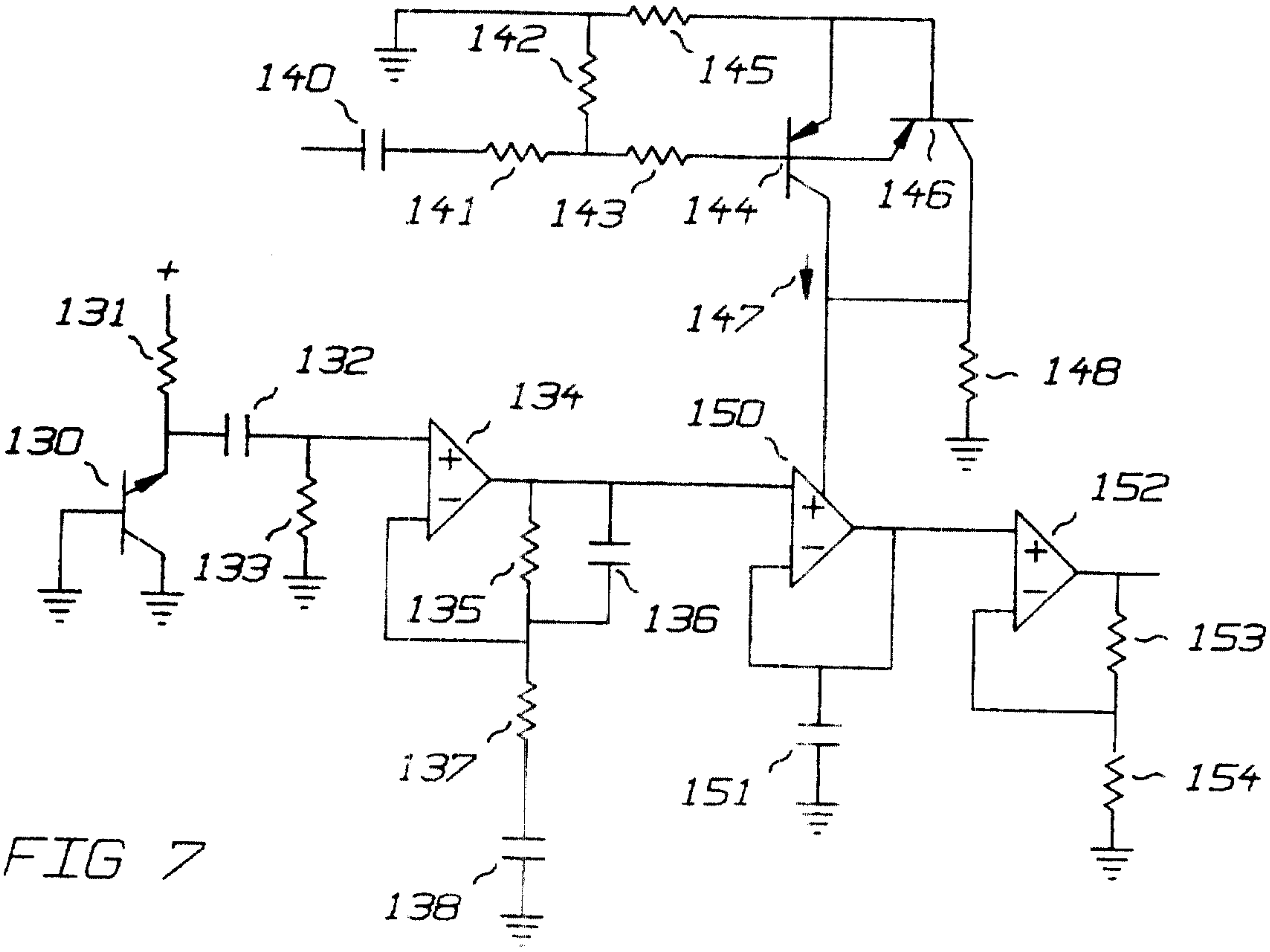
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FIG 7

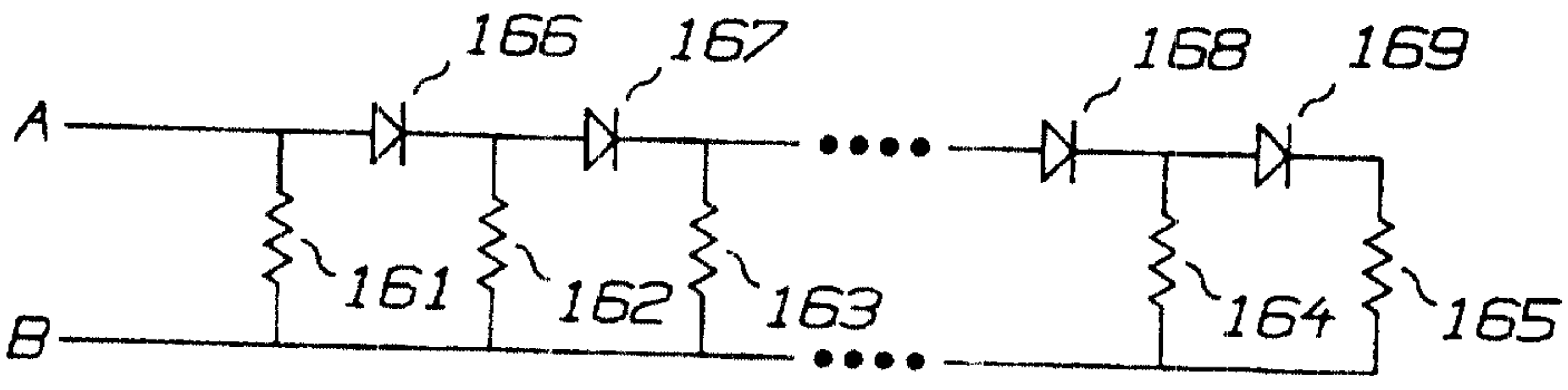

FIG 8

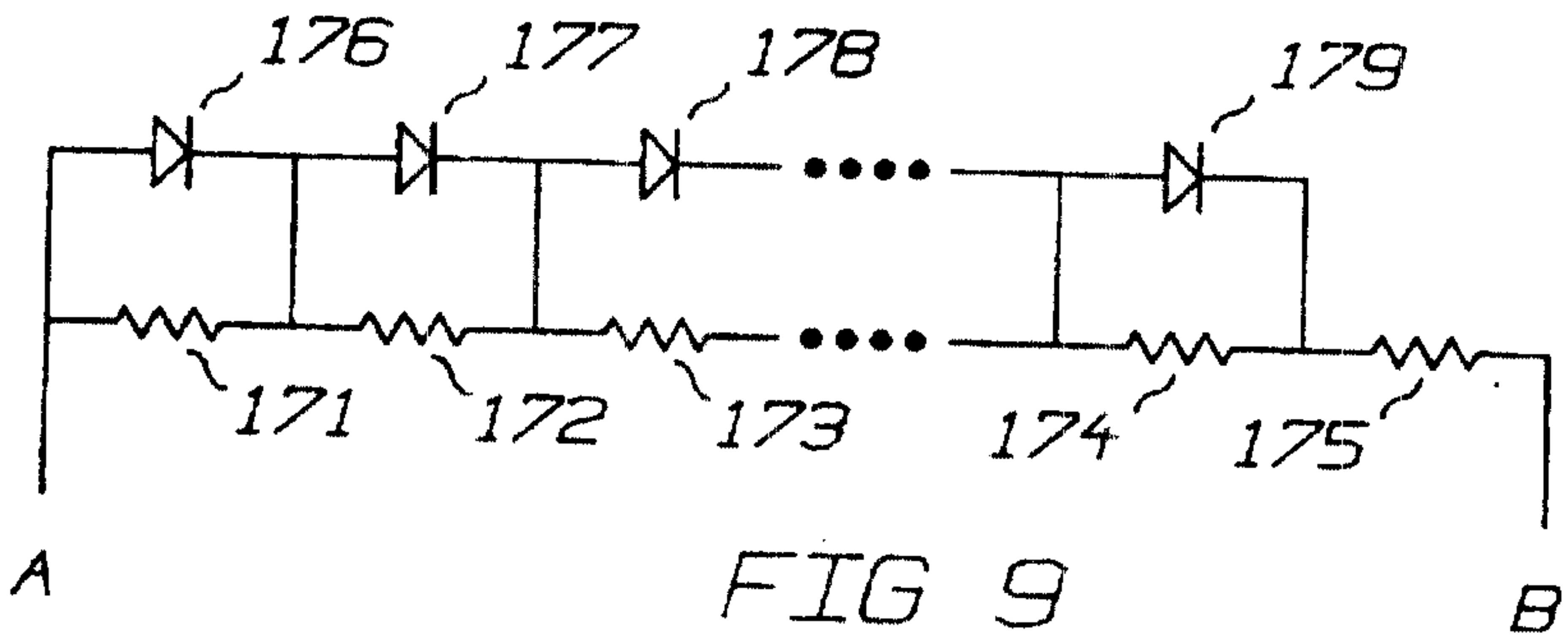

FIG 9

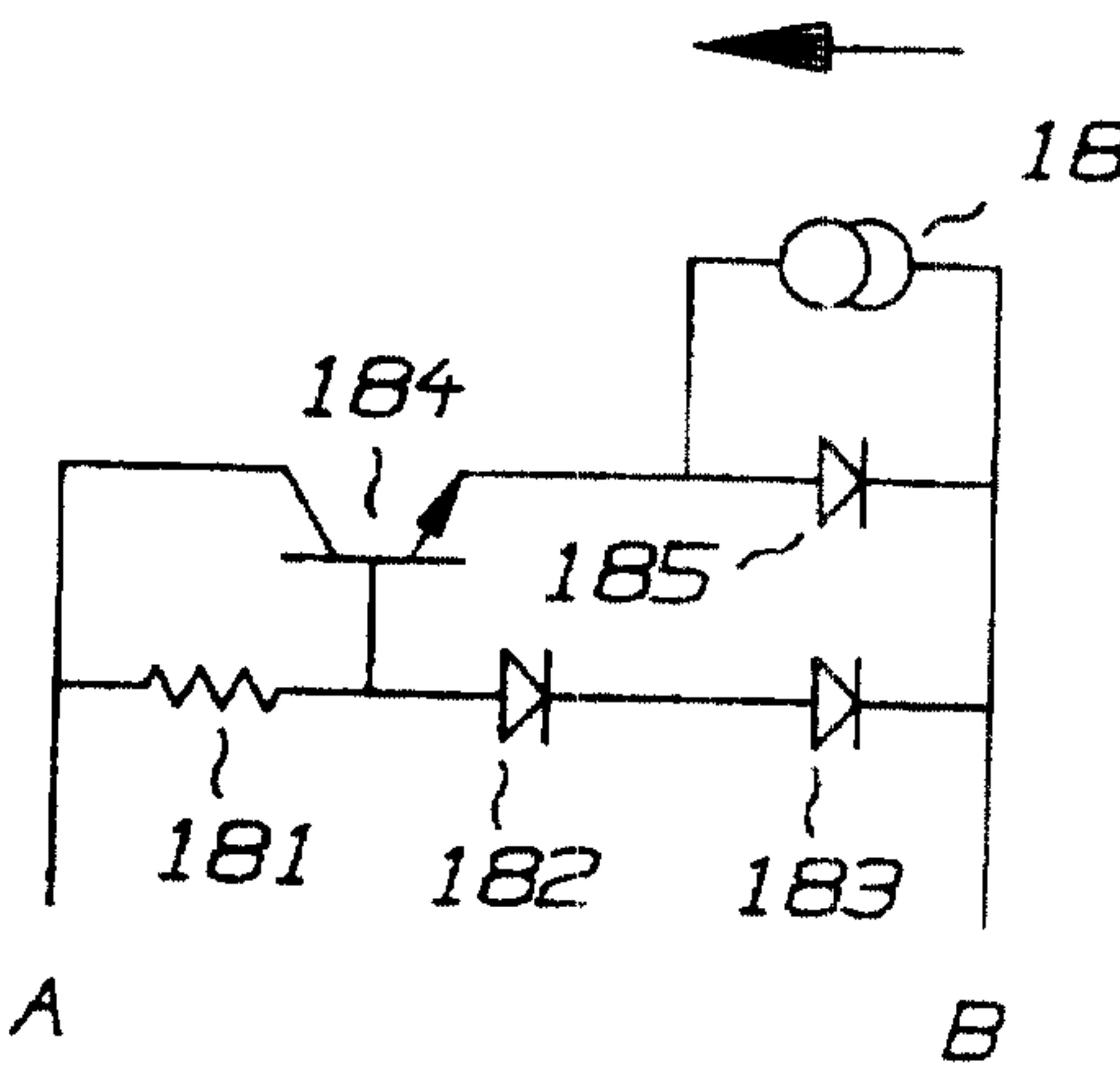

FIG 10

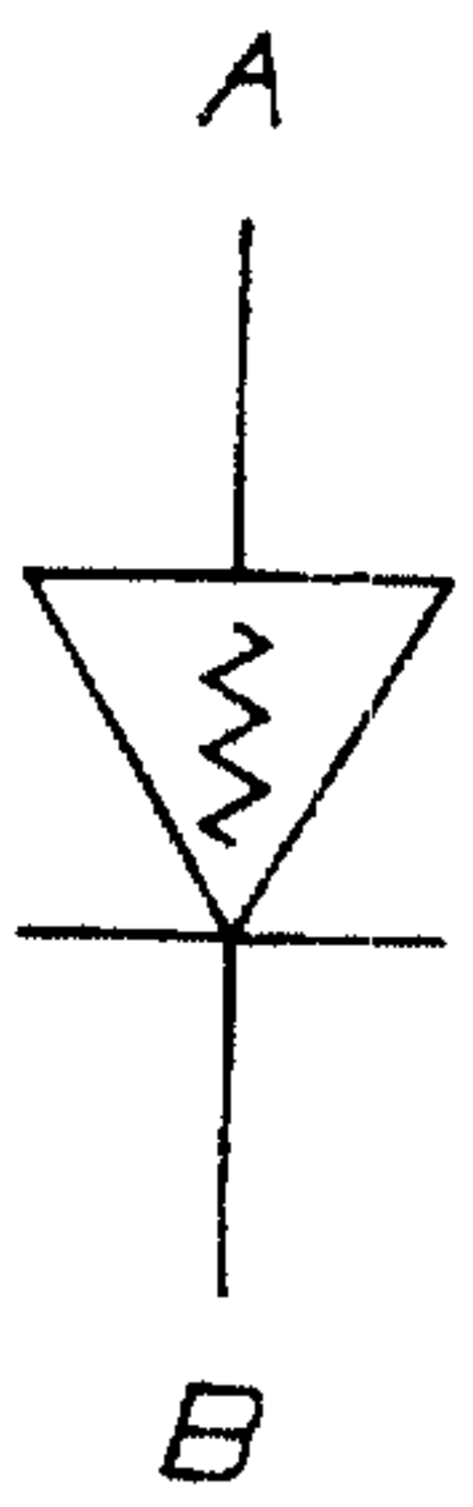

FIG 11

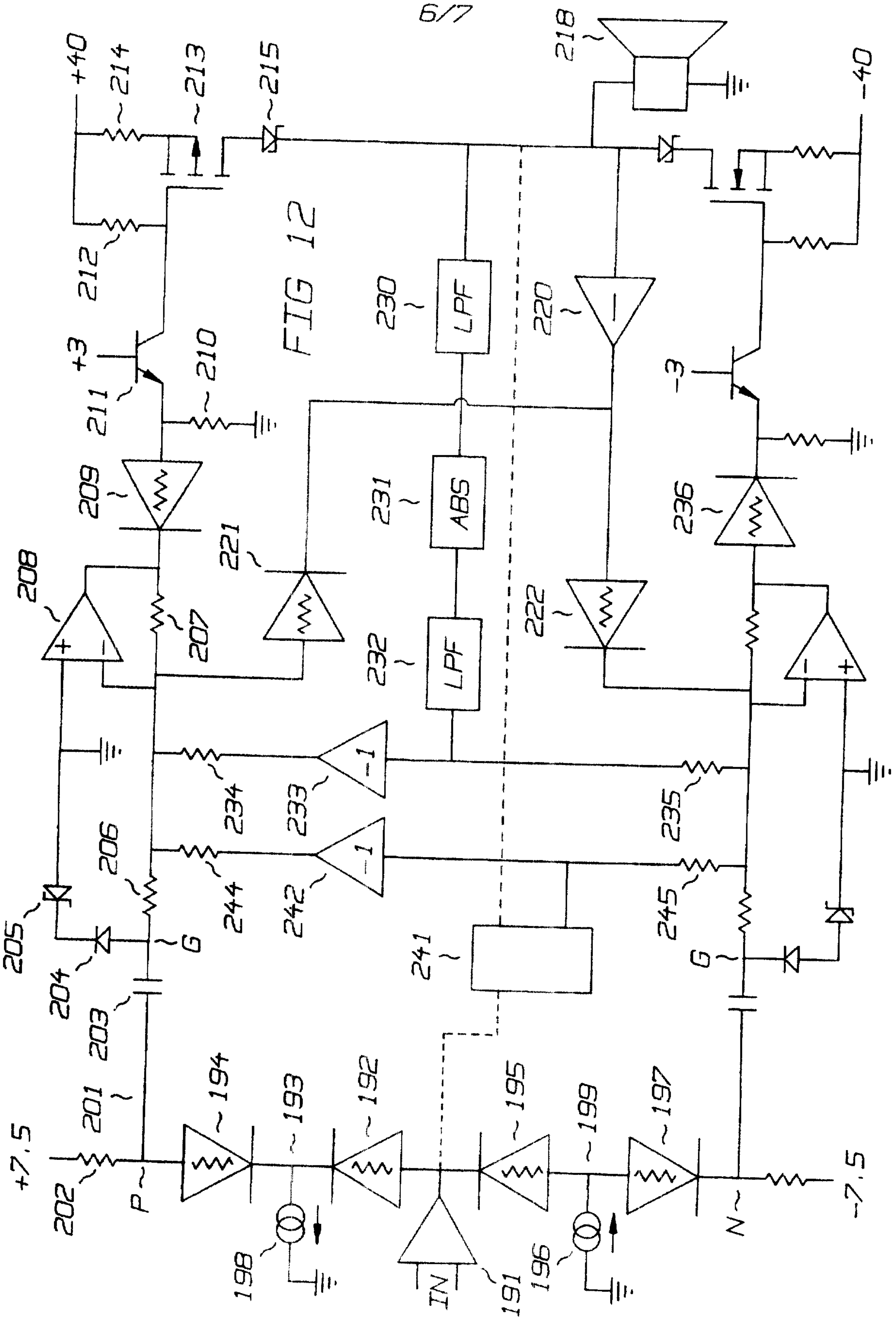
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FIG 12

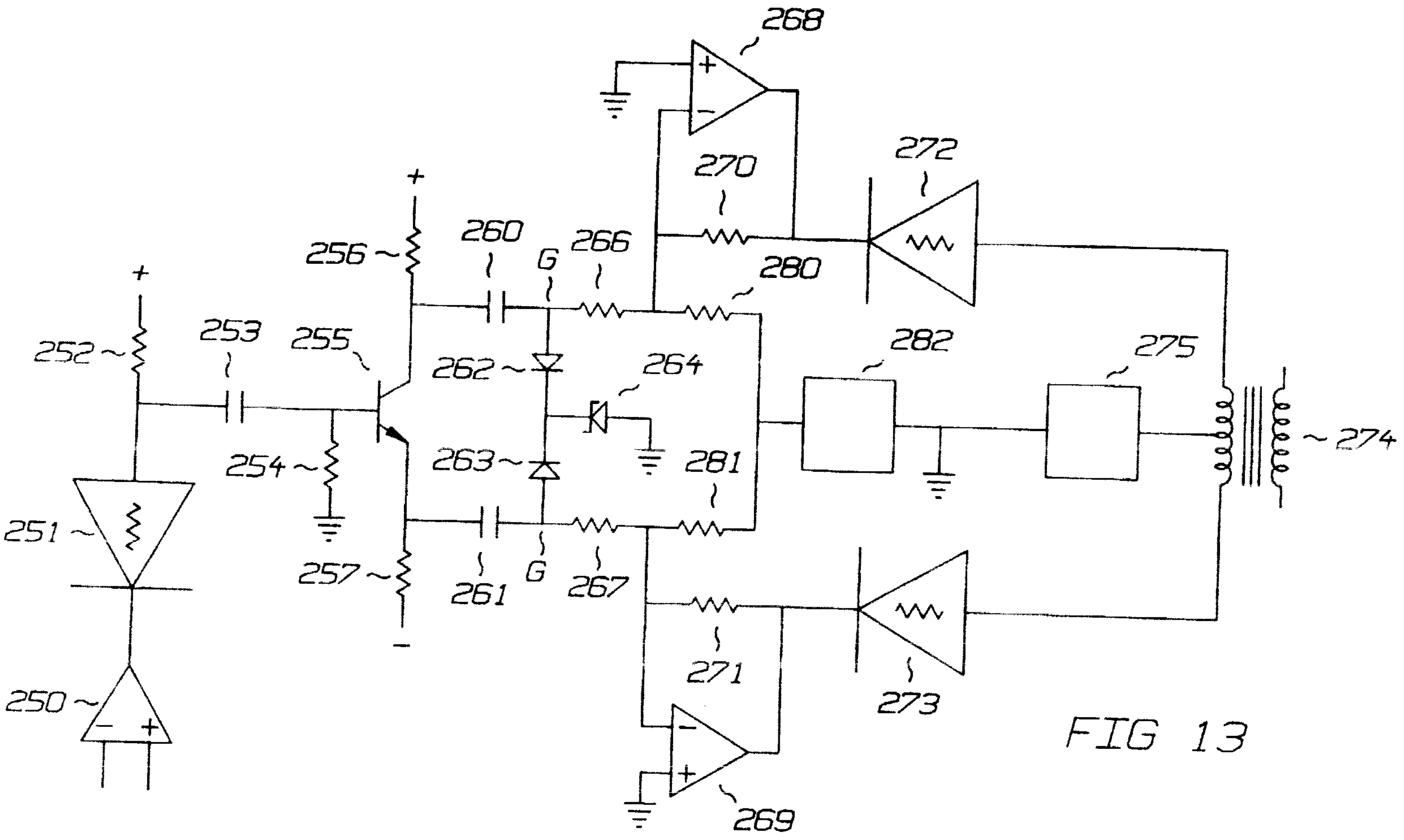


FIG 13

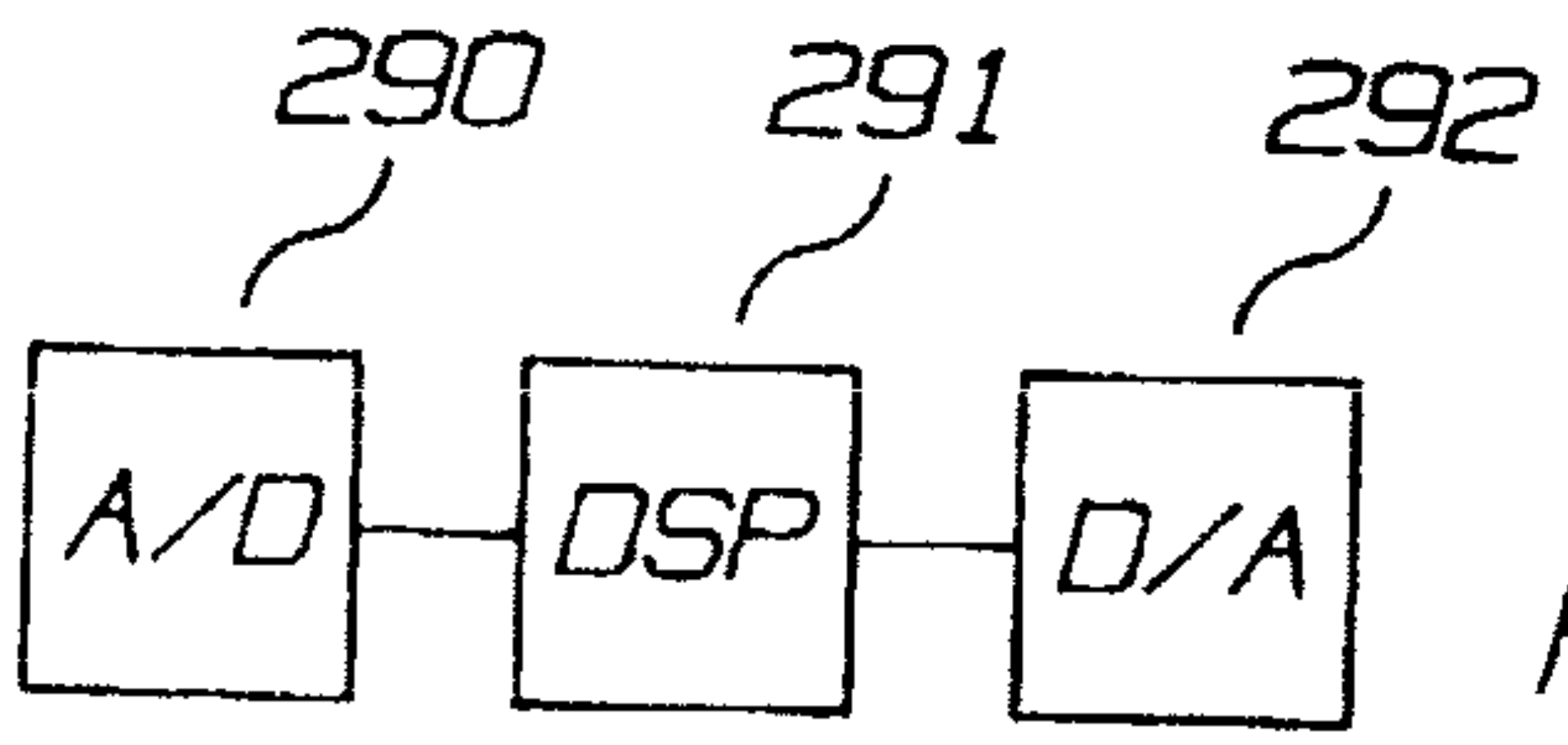


FIG 14

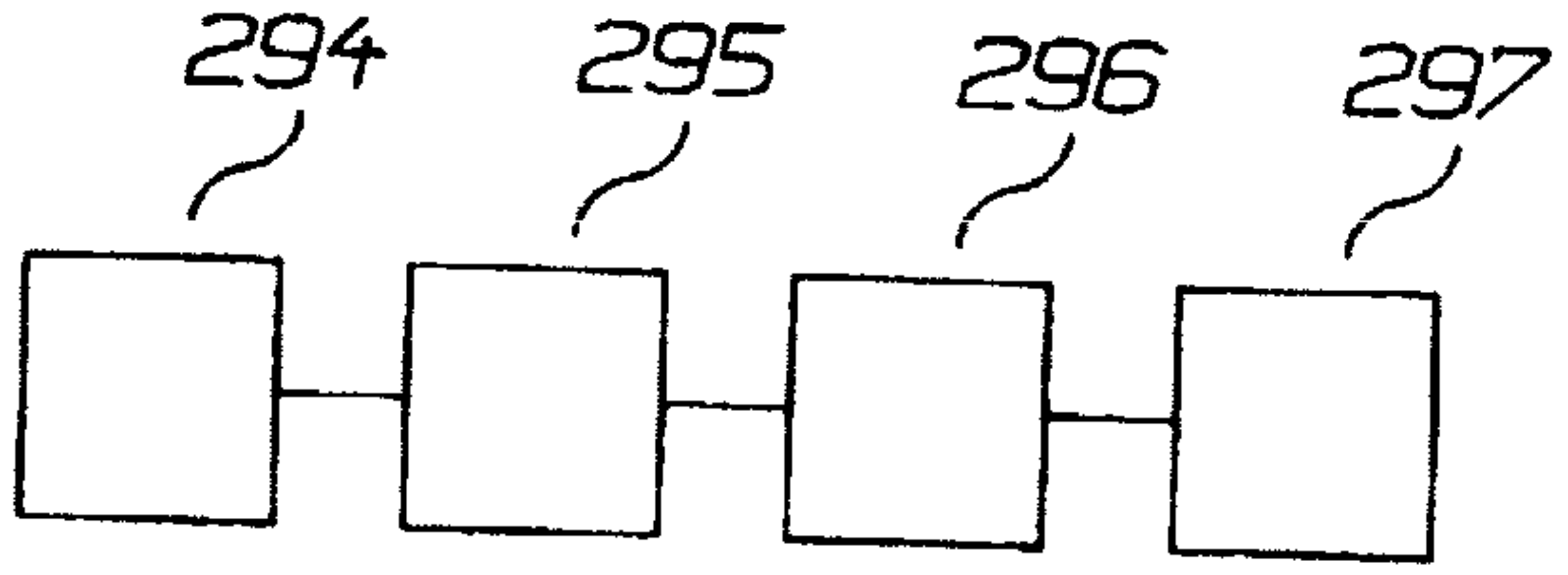
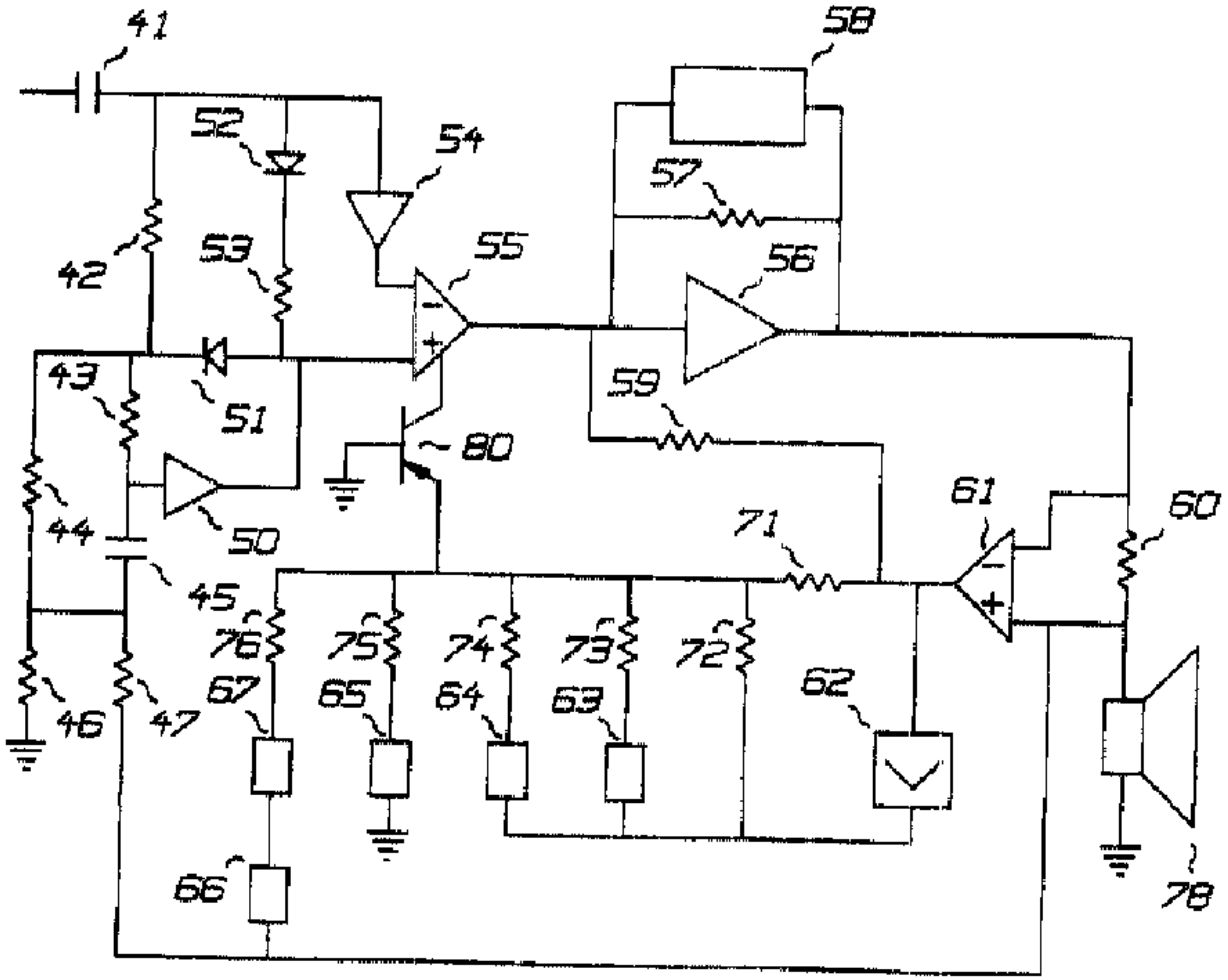


FIG 15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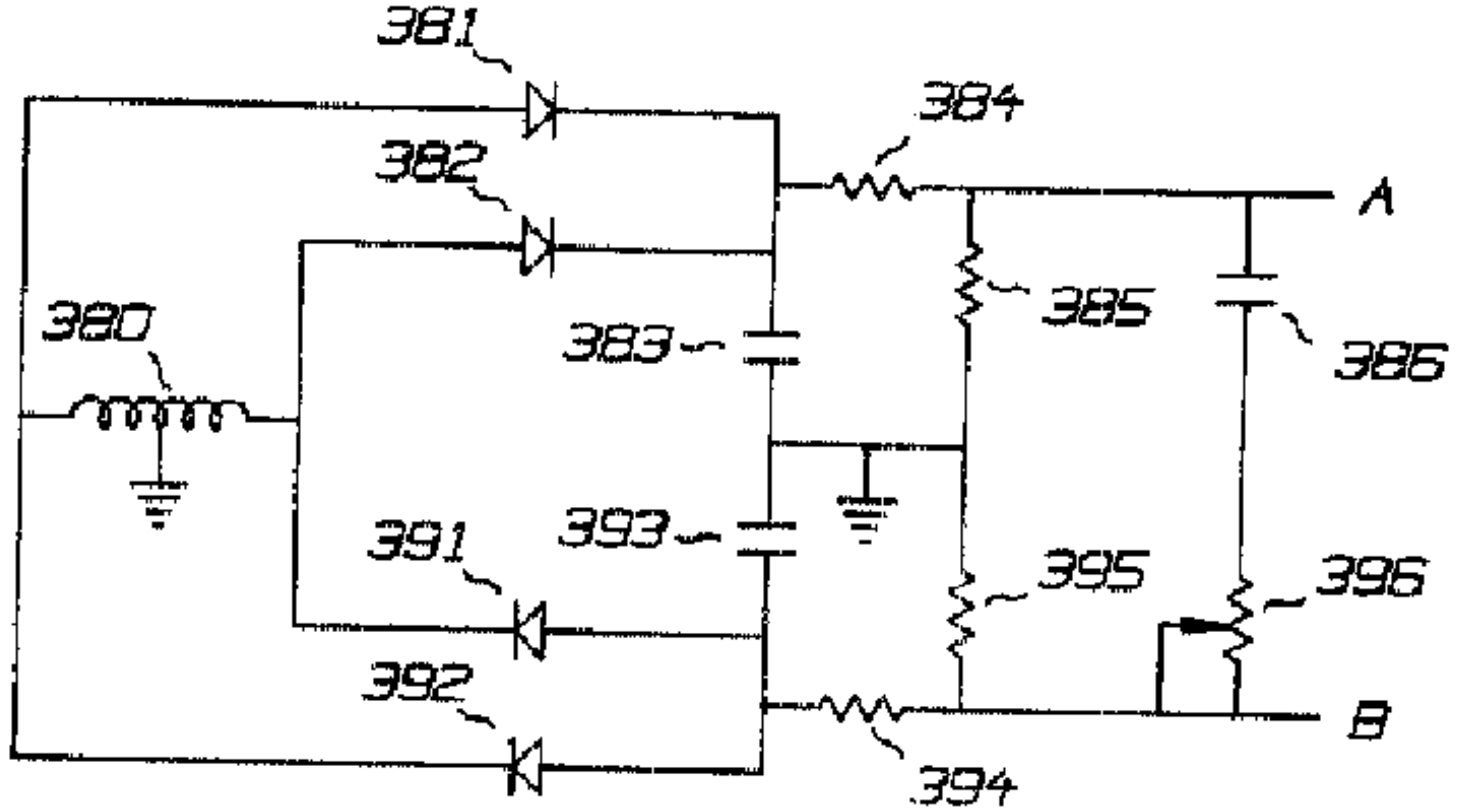
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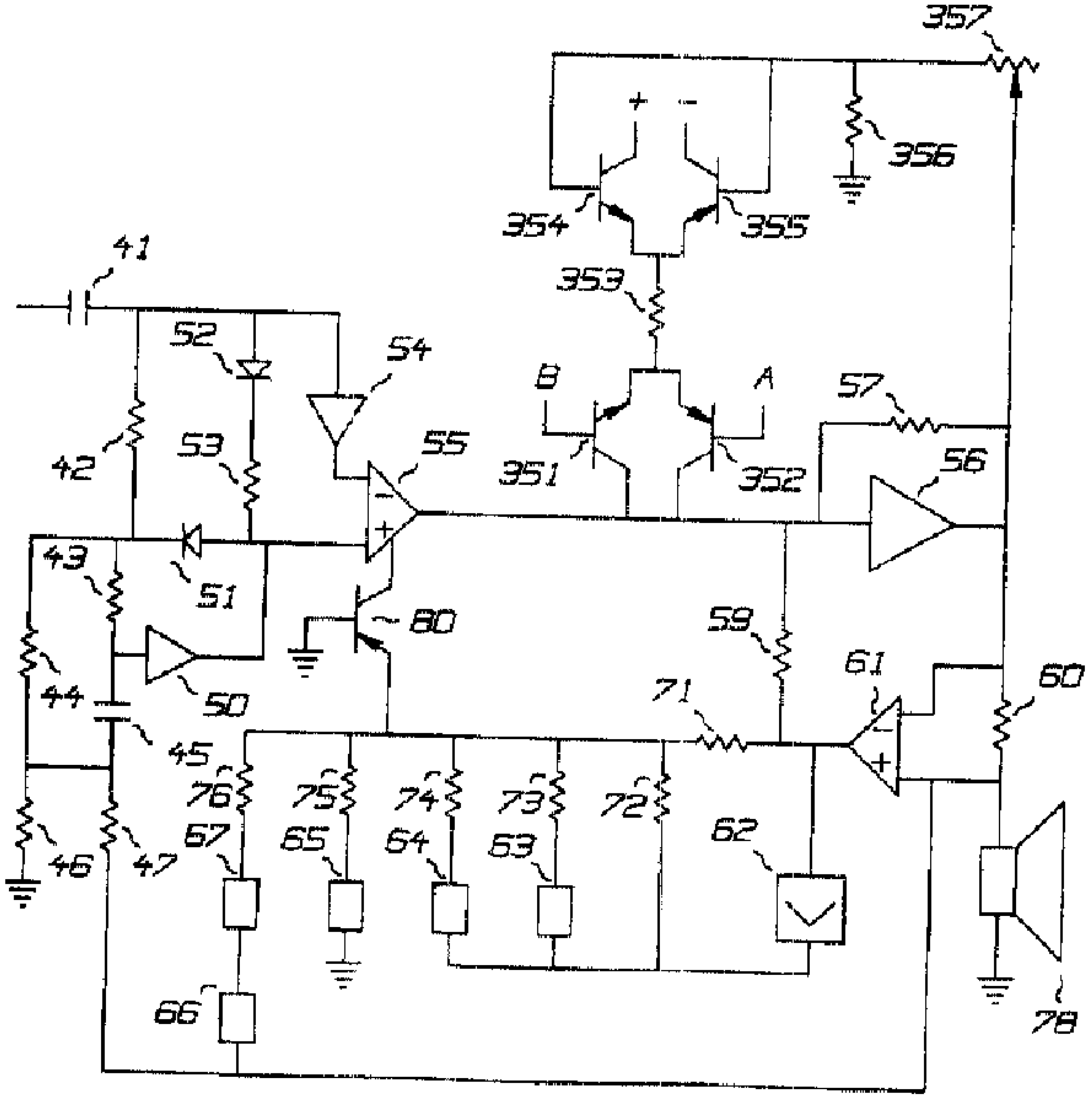
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