

SPECIFICATION

Circuit for suppressing noise caused by scratches on a gramophone record

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The invention relates to a circuit for suppressing audio frequency noise in a signal originating from a gramophone record.

A circuit described in the article "Audio disc scratch filter" by M. R. Sach and J. M. Bullingham, published in "Electronics Letters" December 9 1976, volume 12, No. 25, pages 656-657, comprises a signal processing section arranged between an input and an output, the signal processing section having a delay unit and a noise suppressor arrangement coupled thereto. The circuit also includes a control signal section connected to the input, the control signal section being provided with a noise detector.

In this prior art circuit the signal delayed in the delay unit is passed to the output via the noise suppressor in the absence of noise. However if a pulse-shaped interference, for example caused by a scratch on the gramophone record, is detected in the noise detector, a control signal is generated which signal activates the noise suppressor. During the activation period of the noise suppressor a switching-over is effected in the noise suppressor, and the delayed signal applied to the output is replaced by the non-delayed input signal. This switching-over state, wherein the input is connected to the output via the noise suppressor, is maintained for 70 msec. whereafter a switching back to the original condition automatically takes place, the non-delayed signal being replaced again by the delayed signal. During the period wherein the input is connected to the output via the noise suppressor interferences then occurring cannot be suppressed and are, consequently audible in the audio frequency output signal.

It is an object of the invention to improve the known circuit.

According to the present invention there is provided a circuit for suppressing audio frequency noise in a signal originating from a gramophone record, comprising an input, an output, a signal processing section arranged between the input and the output, the signal processing section having a delay unit and a noise suppressor in the form of a multiplier coupled between the delay unit and the output, the multiplier having a control input, and a control signal section connected to the input, the control signal section having a noise detector which is coupled by a function generator to the control input, the function generator being adapted to generate a control signal at the occurrence of noise which control signal varies the delayed signal originating from the record during a first period of time from a working level to a suppressing level, the signal having a minimum value at this suppressing level, the control signal being maintained during a second period of time at the suppressing level and being varied during a third period of time from the suppressing level to the working level, said periods of time having values between substantially 0.5 msec and 3 msec, the time delay of the delay unit being at least equal to

the first-mentioned period.

From the article "An Estimation of Annoyance Caused by Dropouts in Magnetically Recorded Music" by B. L. Cardozo and G. Domburg, published in "Journal of the Audio Engineering Society", October 1968, volume 16, No. 4, pages 426-429 it is known that for signal interruptions not exceeding a duration of approximately 10msec. both the reduction and the increase in the signal amplitude occurs gradually over a certain period of time and are inaudible, irrespective of the value of the signal amplitude variation.

The invention uses this phenomenon, which is known *per se*, by providing a circuit for producing such signal interruptions artificially and substitutes them for the generally pulse-shaped interferences present in the original signal, which interferences may be produced during playing of a record, for example owing to damage or contamination of the record.

Experiments showed that with the above-mentioned periods of the signal reduction time, the signal suppression time and the signal increase time, such interferences are in practice suppressed to such an extent that they are substantially inaudible.

In an embodiment of a circuit in accordance with the invention the function generator comprises a cascade arrangement of a multivibrator connected to the noise detector, an integrating network for integrating the output pulses of the monostable multivibrator and an amplitude limiter connected to the control input of the noise suppressor for limiting the output pulses of the integrating network to one amplitude value.

A square pulse having a constant duration and amplitude can be obtained by means of the monostable multivibrator at each interference detected by the noise detector. By integrating this pulse in the integrating network and limiting it thereafter in the amplitude limiter a control pulse of a suitable shape is obtained for achieving therewith a signal gain variation in accordance with the variation mentioned above.

In another embodiment of the circuit in accordance with the invention the integrating network comprises an operational amplifier shunted by a capacitor, an input of the amplifier is connected, via a resistor, to an output of the monostable multivibrator, an output is coupled to a common junction of first and second zener diodes included in the amplitude limiter and arranged in series between supply voltage rails.

The present invention will now be described, by way of example, with reference to the accompanying drawings, wherein:

Figure 1 shows an embodiment of a circuit in accordance with the invention,

Figure 1a shows a preferred embodiment of a control pulse for suppressing the stereo signal at the occurrence of noise in the circuit shown in Figure 1,

Figure 2 shows a practical implementation of a full-wave rectifier and a smoothing filter as used in the circuit of Figure 1,

Figure 3 shows a practical implementation of a

converter for converting a pulse frequency into a d.c. voltage as used in the circuit of Figure 1, and

Figure 4 shows another embodiment of a circuit in accordance with the invention wherein the noise suppression is performed digitally on PCM signals representing an audio frequency stereo signal.

In Figure 1 a signal processing section 9 is arranged between inputs 1 and 2 and outputs 3 and 4 and a control signal section 10 is included in a forward control loop. The signal processing section 9 comprises two identical parallel signal paths, each signal path including a cascade arrangement of pre-amplifiers 5 and 6, connected to the inputs 1 and 2, respectively, delay circuits 7a and 7b and controlled amplifiers 8a and 8b functioning as multipliers being connected to the outputs 3 and 4. The delay circuits 7a and 7b and the controlled amplifiers 8a and 8b constitute a delay unit 7 and a noise suppressor 8, respectively.

The control signal section 10 comprises preamplifiers 13 and 14, connected to the inputs 1 and 2, respectively, and a differential amplifier 29 connected to outputs 90 and 91 of the preamplifiers 13 and 14. A sum amplifier 21' of a signal adder circuit is also coupled to the outputs 90 and 91, via resistors 19 and 20, respectively. The differential amplifier 29 is coupled via a differentiator 53 to a difference input 12 of a dynamic compressor 11, the sum amplifier 21' being connected to a sum signal input 15 thereof. Via amplifiers 30 and 35 an output 33 of the dynamic compressor 11 is coupled to a monostable multivibrator 36 which is connected to a converter 37 for converting a pulse frequency into a d.c. voltage. The amplifier 30 is provided with a fixed threshold voltage.

The converter 37 (together with the amplifiers 30, 35 and the monostable circuit 36) is shunted by a series arrangement of a coupling capacitor 38 and a resistor 39 included in a connecting line 17. The output 33 of the dynamic compressor 11 is coupled to an input 40 of a noise detector 32 via the connecting line 17. The converter 37 is also connected to the input 40. An output 18 of the noise detector 32 is coupled to a function generator 34. An output of the function generator 34 is connected to a control input 49 of the noise suppressor 8.

The left-hand and right-hand audio signals (L and R, respectively) of an audio frequency stereo signal recorded on a gramophone record and originating from a scanning element, not shown, of, for example a gramophone, are applied separately to the inputs 1 and 2. In the signal processing section 9 these audio signals are first amplified to the same extent by means of the pre-amplifier 5 and 6, thereafter delayed in the delay unit 7 over an equal period of time and, finally, the noise is removed in the noise suppressor 8 by a signal suppression during the occurrence of noise produced, for example, by a scratch in the gramophone record. The control signal for those signal suppressions is applied to the noise suppressor 8 via the control input 49. This control signal is supplied by the control signal section 10 after detection of noise in the audio signals L and R, applied to the inputs 1 and 2.

Besides amplitude variations owing to the

dynamic range the disturbed stereo signal also shows pulse-shaped amplitude variations superimposed thereon, which are of a considerably shorter duration than the first-mentioned amplitude variations. They may be produced by, for example, scratch noise on the gramophone record and therefore be unwanted. However, they may also derive from musical instruments, such as, for example, trumpets or percussion instruments, in which case they are desired. For convenience of description the desired and undesired pulse-shaped amplitude variations will hereinafter be called music pulses and noise pulses, respectively.

The detection of noise pulses is done in two selection stages.

In the first selection stage the pulse-shaped amplitude variations, that is to say the noise and music pulses, are separated from the amplitude variations caused by the dynamic range of the sound signal. In the second selection stage the noise pulses are separated from the music pulses. For this use is made of the feature that the time intervals between the noise pulses are generally much larger than those between the music pulses.

The first selection stage can be realised by means of, for example, an amplifier having a threshold, which amplifies only those signals whose amplitude exceeds a given threshold voltage. This threshold voltage can be made to vary with a certain slowness with the dynamic range of the stereo signal, so that only the noise and music pulses superimposed thereon exceed the threshold voltage. Alternatively, it is possible to give the threshold voltage a fixed value and to compress the disturbed stereo signal with a certain slowness so that only the noise and music pulses superimposed on the dynamic range exceed this fixed threshold value.

The circuit shown in Figure 1 uses the latter possibility for said first selection, starting from the difference signal L-R formed in the differential amplifier 29. For, the article "Audio disc scratch filter" by M. R. Sach and J. M. Bullingham, published in Electronics Letters dated December 9th, 1976, volume 12, pages 656 and 657 discloses that the difference signal L-R of a stereo signal, originating from a gramophone record, is eminently suitable for the detection of noise caused by scratches on the relevant gramophone record.

In the present circuit this difference signal L-R is first differentiated in the differentiator 53. As a result the amplitude of the music and noise pulses is increased relative to the amplitude variations caused by the dynamic range of the stereo signal. To this end the differentiator has a time constant of 0.14 msec in a practical implementation. Thereafter the difference signal L-R is compressed in a controllable amplifier 16a of the dynamic compressor 11. The control signal required for the compression should be a representation of the dynamic range of the undisturbed stereo signal. An acceptable approximation of the dynamic range of the disturbed stereo signal is found in the sum signal L+R because therein the noise pulses are much smaller relative to the other music signals than in each of the signals L, R and L-R.

the sum signal L+R is formed in the signal adder circuit 21 and applied to a controllable amplifier 16b of the dynamic compressor 11 which controllable amplifier 16b is negatively fed back to a high extent.

5 The negative feedback of the controllable amplifier 16b is constituted by a cascade arrangement consisting of an amplifier stage 23, a full-wave rectifier 24 and a smoothing filter 25, arranged between an output 22 and a control input 28. The time constant of the smoothing filter 25 is chosen to be so high that the brief pulse-shaped amplitude variations, that is to say the noise and music pulses, cannot be followed. The control signal applied to the control input 28 consequently represents fairly accurately the dynamic range of the stereo signal and controls, with an increasing or decreasing amplitude of the sum signal, the gain of the two amplifiers 16a and 16b to a smaller and larger gain, respectively.

As a result both the loud and the soft passages in the difference signal L-R are adjusted to substantially the same amplitude level.

Because the noise and music pulses are substantially not present in the control signal they are not eliminated from the difference signal L-R and are consequently maintained.

After this dynamic compression the difference signal L-R is passed to an amplifier 30, having a threshold, wherein said first selection of the music and the noise pulses from the difference signal L-R is effected. To this end the threshold voltage of the amplifier 30 is chosen so that only music and noise pulses can exceed this threshold voltage. They are amplified in an amplifier 35 and thereafter activate the monostable multivibrator 36.

For the second selection, namely the distinction between the noise and the music pulses, the starting point is the known fact that in general noise pulses occur incidentally, that is randomly and generally singularly and music pulses in series of five pulses or more with a frequency of approximately 120 Hz or higher. So the pulse frequency of the output pulses of the monostable multivibrator 36 contains information about the pulse type. This pulse frequency is converted in the converter 37 into a negative d.c. voltage. In a practical implementation the converter 37 has such a charge time constant that the d.c. voltage obtains a maximum negative value with a pulse series of 5 pulses or more within a time of 20 msec and such a discharge time constant that the d.c. voltage exponentially decreases from the maximum value to a minimum value during 200 msec when pulses are absent.

Like the first selection also the second selection can be realised by means of an amplifier having a threshold, which only amplifies those signals whose amplitude exceeds a given threshold voltage. This threshold voltage may be given a fixed value. In that case it is possible to first increase considerably, in a controllable amplifier, using the d.c. voltage of the converter 37 as a control signal, the noise pulses in the output signal of the dynamic compressor 11 of the monostable multivibrator 36, relative to the music pulses occurring in this output signal and to pass thereafter these noise and music pulses to the amplifier having a fixed threshold. The threshold

voltage of the amplifier having a threshold which should be chosen to have a value between the amplitude value of the noise pulses and that of the music pulses.

Alternatively, it is possible to have the threshold voltage vary with the d.c. voltage of the converter 37 and to apply the noise and music pulses such as they occur in the output signal of the dynamic compressor 11 or the monostable multivibrator 36 directly to this amplifier having a variable threshold. This selection method is used in the present circuit. The d.c. voltage of the converter 37 is applied via the input 40 of the noise detector 32 to the base of a transistor 41 included in the noise detector 32 and functioning as an amplifier having a threshold. At the same time the difference signal L-R of the output of the dynamic compressor 11 is applied to the input 40 via the connecting line 17. The emitter of the transistor 41 is connected to ground, its collector is connected to a collector resistor on the one hand and to an output 18 on the other hand.

The threshold voltage of the transistor 41 is determined by its base-emitter bias voltage, which is increased if the negative value of the d.c. voltage of the converter 37 increases. Music pulses which are applied via the connecting line 17 to the base of the transistor 41 produce such a negative d.c. voltage at the output of the converter 37 that they cannot render transistor 41 conductive. Noise pulses on the contrary produce substantially no negative d.c. voltage at the output of the converter 37 so that they can render the transistor 41 conductive via the connecting line 37.

However, noise pulses may also occur during a series of music pulses. Such noise pulses do not render transistor 41 conductive and are, therefore, recognised as desired music pulses. In practice these noise pulses are hardly audible during such music signals. The noise pulses at the output 18 of the noise detector 32 initiate a monostable multivibrator 43 of the function generator 34. The uniform square output pulses of this monostable multivibrator 43 are integrated in an integrating network 50 consisting of a series arrangement of a variable resistor 44 connected to the monostable multivibrator 43 and an operational amplifier 45 shunted by a capacitance 46 into triangular pulses. The slope thereof is determined by the time constant of the variable resistor 44 and the capacitance 46. The amplitude of these triangular pulses is limited in a limiter circuit 51 of the function generator 34, constituted by a series arrangement of zener diodes 47 and 48. The common junction of these zener diodes is connected to both an output of the integrating network 50 and the control input 49 of the noise suppressor 8. Figure 1a shows the shape of the control pulses, wherein the period of time in which the trajectories, a, b and c are passed may amount to 1 to 2 msec in practice. A maximum suppression of the stereo signal is effected in noise suppressor 8 during the trajectory b. During the trajectories a and c, respectively, the gain of the stereo signal is reduced and increased, respectively, in the controlled amplifiers 8a and 8b of the noise suppressor 8.

The duration of the delay realised in the delay unit

7 should be at least equal to the period elapsing between the detection of noise and the occurrence of the trajectory *b* in the control pulse derived therefrom.

- 5 In a circuit shown in Figure 1, tested in practice, the amplifiers 5, 6, 13, 14, 21', 29, 23, 35 and 45 were provided with operational amplifiers of the type TCA 680, the amplifier 30 with a BC 550-type transistor and the monostable multivibrators 36 and 43 with a
10 HEF 4528-type integrated circuit. The delay unit 7 comprised an integrated, digital, clock-controlled, dual charge transfer store (bucket-brigade delay line) of the type TDA 1022, with which a signal delay per channel of between 3 and 1.5 msec could be realised
15 under the control of a clock frequency adjustable between 85 KHz and 170 KHz. The noise suppressor 8 as well as the controllable amplifiers 16a and 16b were implemented by means of a type TCA 730 integrated dual controllable amplifier. The transistor 41
20 is of the type BC 550. A 2 msec pulse width was chosen for the monostable multivibrator 36, a 5 msec pulse width for the monostable multivibrator 43.

The integrating network 50 and the limiter circuit
25 51 were dimensioned so that the trajectories *a* and *c* of the control pulse depicted in Figure 1a were approximately 1.5 msec. The delay of delay circuit 7 was also adjusted to 1.5 msec.

Figure 2 shows a detailed preferred embodiment
30 of the full-wave rectifier 24 and the smoothing filter 25 wherein elements corresponding to the elements of the circuit shown in Figure 1 have been given the same reference numerals.

The full-wave rectifier 24 comprises operational
35 amplifiers 61 and 62 connected by means of their non-inverting and inverting input, respectively, to a signal input 31 via matching resistors 63 and 64. The operational amplifiers 61 and 62 are both negatively fed back from their outputs to the respective inverting
40 inputs via diodes 65 and 67 respectively. The cathodes of diodes 65 and 67 are connected to said inverting inputs on the one hand and on the other hand to an input 71 of smoothing filter 25 via resistors 69 and 70. The outputs of the operational ampli-
45 fiers 61 and 62 are connected to the cathodes of the diodes 66 and 68, respectively, through which they are coupled to the input 71. The non-inverting input of the operational amplifier 62 is connected to a reference voltage.

50 The smoothing filter 25 comprises a series arrangement of a resistor 26 and a smoothing capacitor 27, arranged between the input 71 and ground. The connection between resistor 26 and smoothing capacitor 27 is coupled to the control
55 input 28.

In the case of a positive polarity of the signal at the signal input 31 relative to the reference voltage at the non-inverting input of the operational amplifier 62 diode 65 is conducting, so that the operational amp-
60 lifier 61 is negatively fed back to a great extent and its gain is negligibly small. This positive signal appears amplified and inverted, at the output of the operational amplifier 62. Diode 67 is cut-off and the output voltage, that is to say the inverted positive
65 input signal of the signal input 31 appears at the

input 71 of smoothing filter 25 via the conducting diode 68. Diode 66 is also cut-off so that the output impedance of the operational amplifier 61 no longer loads the output of the operational amplifier 62.

- 70 In the case of a negative polarity of the signal at the signal input 31 relative to said reference voltage diode 65 is cut-off and diode 66 conducts. The output signal of the operational amplifier 61, that is to say the negative input signal of the signal input 31
75 appears, amplified, at the input 71 of smoothing filter 25 via the conducting diode 66. The diode 67 is then conducting so that the operational amplifier 62 is negatively fed back to a great extent and its amplification is negligibly small. Diode 68 is cut-off so
80 that the output impedance of the operational amplifier 62 does not load the output of the operational amplifier 61.

So a signal whose amplitude is equal to the difference between the reference voltage at the non-
85 inverting input of the operational amplifier 62 and the amplitude of the full-wave rectified sum signal $L+R$ appears at the input 71 of smoothing filter 25. This signal increases when the amplitude of the sum signal decreases and *vice versa*. In the smoothing
90 filter the signal is smoothed to a control signal for the controllable amplifiers 16a and 16b, for which the smoothing filter has a time constant of approximately 3.3 msec. With such a time constant the control
95 signal only represents amplitude variations owing to the dynamic range of the stereo signal and substantially not the noise and musical pulses.

In a practical implementation the operational amp-
100 lifiers 61 and 62 were integrated and of the type TCA 680; the diodes of the type BAX 13. Reference voltage was 8 volts. The values of the resistors 63, 64, 69 and 70 were 10 K Ω , the value of the resistor 26 was 3K3 Ω while the smoothing capacitor 27 had a capacitance of 1 μ F.

Figure 3 shows a detailed preferred embodiment
105 of the converter 37 arranged between an output 80 of the monostable multivibrator 36 and the input 40 of the noise detector 32.

The converter 37 comprises a series arrangement of a coupling capacitor 81, a resistor 83 and a diode
110 84 whose cathode is connected to the resistor 83. The anode of the diode 84 is connected to ground through a parallel RC member consisting of a capacitor 85 and a potentiometer 86. The wiper of potentiometer 86 is coupled via a matching resistor
115 87 to the input 40 of the noise detector 32. The connection between the coupling capacitor 81 and resistor 83 is connected to ground via a matching resistor 82.

The negative pulses at the output 80 of the mono-
120 stable multivibrator 36 charge the capacitor 85 negatively through the diode 84 which is conductive for these pulses. This capacitor 85 discharges via the body of potentiometer 86, the RC time constant being approximately 0.1 sec in a practical implemen-
125 tation. The RC member was rated so that for a pulse train of 5 to 6 pulses a maximum negative capacitor voltage of approximately -2.2 V is achieved within 20 msec. An adjustable portion of the capacitor vol-
130 tage is tapped-off at the wiper of the potentiometer 86 and applied as a varying threshold voltage to the

input 40 of noise detector 32 via matching resistor 87. The wiper was adjusted so that the threshold voltage of the noise detector achieved a minimum value, thus increasing the sensitivity of the noise detector to a maximum value after no pulses had occurred for 200 msec.

In this practical implementation the capacitors 81 and 85 had a capacitance value of $22\mu\text{F}$ and $1\mu\text{F}$, respectively. The resistors 82, 83 and 87 had a value of $39\text{K}\Omega$, $5.1\text{K}\Omega$ and $39\text{K}\Omega$, respectively. The potentiometer 86 had a maximum value of $100\text{K}\Omega$. Diode 84 was of the type BAX 13.

Figure 4 shows another embodiment of a circuit in accordance with the invention in which the sub-circuits whose function correspond to that of the circuit shown in Figure 1 have been given the same reference numerals.

A digital signal is applied to the input 1. This digital signal may, for example, consist of a sequence of digital code words representing number values, corresponding to sampling values of an analog audio frequency signal. In the case of a stereo signal these code words may, for example, alternately indicate sampling values of the left-hand and right-hand audio signal.

Interferences in this digital signal which are audible after a digital-to-analog conversion can be suppressed inaudibly by multiplying, in case such an interference is detected, the number values of the code words in a multiplier 8, which functions as a noise suppressor, by weighting factors which decrease during said first periods in numerical value from 1 to a low value, for example 0, are maintained during the second period at this low numerical value and increase during the third period to the value 1.

Such weighting factors can be generated in a simple manner in the function generator 34, for example by means of an ROM (read only memory), which is read under the control of a clock at each interference.

To enable a simple detection of disturbed code words, one or more check bits are added to each code word during coding of the sampling values. The values of these check bits are determined by the values of the data bits and that in such a manner that, for example, the total number of ones in each code word and its check bits remains constant.

By now counting, for example, the plurality of ones in each code word and the associated check bits in an adder 32, which functions as a noise detector, the disturbed code words can be recognized. If the plurality of consecutive disturbed code words exceeds a given number, so that they become audible as noise after a digital-to-analog conversion, the function generator 34 is activated and noise suppression is performed in multiplier 8.

The delay unit 7 may be provided in known manner with shift registers. Then the delay time should be at least equal to the time during which last-mentioned plurality of code words occurs and to said first period of time.

CLAIMS

1. A circuit for suppressing audio frequency noise in a signal originating from a gramophone record, comprising an input, an output, a signal processing section arranged between the input and the

output, the signal processing section having a delay unit and a noise suppressor in the form of a multiplier coupled between the delay unit and the output, the multiplier having a control input, and a control signal section connected to the input, the control signal section having a noise detector which is coupled by a function generator to the control input, the function generator being adapted to generate a control signal at the occurrence of noise which control signal varies the delayed signal originating from the record during a first period of time from a working level to a suppressing level, the signal having a minimum value at this suppressing level, the control signal being maintained during a second period of time at the suppressing level and being varied during a third period of time from the suppressing level to the working level, said periods of time having values between substantially 0.5 msec and 3 msec, the time delay of the delay unit being at least equal to the first-mentioned period.

2. A circuit as claimed in Claim 1, wherein the function generator comprises a cascade circuit of a monostable multivibrator connected to the noise detector an integrating network for integrating the output pulses of the monostable multivibrator and an amplitude limiter connected to the control input of the noise suppressor, for limiting the output pulses of the integrating network to one amplitude value.

3. A circuit as claimed in Claim 2, wherein the integrating network comprises an operational amplifier, shunted by a capacitor having an input connected via a resistor to an output of the monostable multivibrator and an output coupled to a common junction of first and second zener diodes included in the amplitude limiter and arranged in series between supply voltage rails.

4. A circuit for suppressing audio frequency noise in a signal originating from, a gramophone record as claimed in Claim 1, constructed and arranged to operate substantially as hereinbefore described with reference to and as shown in Figures 1 to 3 of the accompanying drawings.

5. A circuit for suppressing audio frequency noise in a signal originating from, a gramophone record, constructed and arranged to operate substantially as hereinbefore described with reference to and as shown in Figure 4 of the accompanying drawings.

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Interferences in this digital signal which are audible after a digital-to-analog conversion can be suppressed inaudibly by multiplying, in case such an interference is detected, the number values of the code words in a multiplier 8, which functions as a noise suppressor, by weighting factors which decrease during said first periods in numerical value from 1 to a low value, for example 0, are maintained during the second period at this low numerical value and increase during the third period to the value 1.

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CLAIMS

1. A circuit for suppressing audio frequency noise in a signal originating from a gramophone record, comprising an input, an output, a signal processing section arranged between the input and the

output, the signal processing section having a delay unit and a noise suppressor in the form of a multiplier coupled between the delay unit and the output, the multiplier having a control input, and a control signal section connected to the input, the control signal section having a noise detector which is coupled by a function generator to the control input, the function generator being adapted to generate a control signal at the occurrence of noise which control signal varies the delayed signal originating from the record during a first period of time from a working level to a suppressing level, the signal having a minimum value at this suppressing level, the control signal being maintained during a second period of time at the suppressing level and being varied during a third period of time from the suppressing level to the working level, said periods of time having values between substantially 0.5 msec and 3 msec, the time delay of the delay unit being at least equal to the first-mentioned period.

2. A circuit as claimed in Claim 1, wherein the function generator comprises a cascade circuit of a monostable multivibrator connected to the noise detector an integrating network for integrating the output pulses of the monostable multivibrator and an amplitude limiter connected to the control input of the noise suppressor, for limiting the output pulses of the integrating network to one amplitude value.

3. A circuit as claimed in Claim 2, wherein the integrating network comprises an operational amplifier, shunted by a capacitor having an input connected via a resistor to an output of the monostable multivibrator and an output coupled to a common junction of first and second zener diodes included in the amplitude limiter and arranged in series between supply voltage rails.

4. A circuit for suppressing audio frequency noise in a signal originating from, a gramophone record as claimed in Claim 1, constructed and arranged to operate substantially as hereinbefore described with reference to and as shown in Figures 1 to 3 of the accompanying drawings.

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