

The invention relates to a method for the digital compensation of a predetermined interference signal in accordance with the first part of the main claim.

A method in accordance with the first part of the main claim is known from IEEE Transactions on Biomedical Engineering, Vol BME-30, No 4, April 1983, pages 234 - 235, New York, USA; G.S. Furno et al.: "A learning filter for removing noise interference" in which the compensating signal is obtained by sampling the signal suffering from interference during a single period of the interference signal in which solely the interference signal, but not however the useful signal, is present.

A method is known from US-A-4 251 831 in which the useful signal suffering from interference is sampled and a plurality of sampling values are so averaged on each occasion that the interference signal is eliminated by the averaging.

In analogue audio HiFi equipment, in public address systems and also in digitally operative audio components which are based on the analogue to digital conversion of an analogue input signal, problems can arise from (mains voltage) in the analogue circuit sections or from the coupling of the analogue input/output signals to the mains voltage (so-called hum-loops). The problem of the "hum" which is audible in sound signals as well as visible in the picture becomes serious with an increasing number of interconnected individual components and with the rising demand for quality as regards isolation from extraneous voltages or as regards signal dynamics.

In modern devices for the transmission of sound and pictures in the professional field as well as in the consumer field, one tries to avoid stray hum pick-up as far as possible by suitable circuit layout and matching construction of the devices. Nevertheless, in practice, the occurrence of stray hum pick-up - especially in audio systems - cannot basically be avoided as the user himself frequently creates stray hum pick-up by connecting together different devices (which in themselves are free of hum) into a complete system, especially via so-called hum-loops in the interconnecting cables. As a non-expert, the user is generally neither in a position to recognise the earth loop as the cause of the stray hum pick-up nor to get rid of it by selective interruption or alteration of the earth connections.

The object of the invention is to conceive a "self hum-eliminating audio device". An appropriate circuit can make electronic devices, especially in the field of home

entertainment, tolerant of errors on the part of the user as regards wiring errors which produce hum.

This object is achieved by the invention defined in claim 1. Advantageous developments and embodiments of the invention are specified in the appendant claims.

The digitally operative hum compensating circuit in accordance with the invention determines the signal components contained in the input signal which are coupled in terms of phase and frequency with the mains frequency, by a signal averaging process. In accordance with the complexity of the circuit, which basically means, in dependence on the sampling frequency of the useful signal ($e(t)$) suffering from interference, as many harmonics of the mains frequency as desired can be taken into account. The compensating signal produced ($k(t)$) is subtracted from the useful signal ($e(t)$) suffering from interference, automatically with the correct magnitude and frequency, whereby, in accordance with the complexity of the circuit, the hum components contained therein are reduced or even completely eliminated.

The invention will be explained hereinafter with the help of the drawings. These show, in:

Fig. 1 a basic circuit for use on analogue signals containing hum components,

Fig. 2 a basic circuit for use on digital signals containing hum components

Fig. 1 shows the basic construction for the generation of a compensating signal ($k_a(t)$) from a useful analogue signal ($e_a(t)$) suffering from interference which is composed of the useful signal ($s_a(t)$) and the interference signal ($n_a(t)$). The output signal ($a_a(t)$) which has been freed of interference, or which has reduced interference, is then present at the output of the basic circuit. The subscript "a" in the Fig. 1 indicates the analogue nature of the signals.

The useful analogue signal ($e_a(t)$) suffering from interference is supplied to a low pass filter (1) as well as to a differential amplifier (9). An A/D converter (2), whose outputs lead to the inputs of a digital adder (4), is connected via a switch (10) actuatable by a control circuit (20) to the low pass filter (1). The outputs of the digital adder (4) are connected to the inputs of a shift register (6) as well as to the inputs of a calculating unit (7). The outputs of the shift register (6)

lead to further inputs of the adder (4). The outputs of the calculating unit (7) extend via a D/A converter (8) to a negative input of the differential amplifier (9), at whose positive input, the useful signal ($e_a(t)$) suffering from interference is present. An n th multiple frequency is generated from the interference creating signal ($f_a(t)$) in a frequency multiplier (3). A cycle rate counter (5) counts the cycles of the interference creating signal ($f(t)$) and passes on the result to the calculating element (7). The A/D converter (2), the digital adder (4), the shift register (6) and the calculating element (7) are clocked with the clock pulses (T) generated by the frequency multiplier (3). The control circuit (20) is started at the commencement of a test run by the mains switch (22) and provides a start signal to the cycle rate counter (5) as well as an enable signal to the calculating element (7) and a reset signal to the shift register (6).

The manner of operation of the arrangement according to Fig. 1 is as follows: In the component chain (1 - 8), the useful signal ($e_a(t)$) suffering from interference is sampled after passage through the low pass filter (1) in the A/D converter (2). The sampling is effected with an integral multiple n of the interference creating signal ($f_a(t)$), which may for example be the mains frequency in the event that a so-called mains induced hum is to be compensated. Each N -bit long sample value is added to the value read out from the last storage position of the shift register (6) having n storage positions. The resulting sum value is written back into the first storage position of the shift register (6) which is likewise operating at the sampling rate. Through these measures, a running addition of a sampling value, each N -bits long, to the already accumulated intermediate sum of the preceding N -bit long sampling values with the appertaining phase relationship (relative to the interference signal ($f_a(t)$)) is realised. After the passage of m sampling periods, each individual summation value circulating in the shift register (6) is then divided in the calculating element (7) by the number m of the N -bit long sampling values already added together in each summation value. As a result, one then obtains the average signal sampling value appertaining to each sampling phase - relative to the signal ($f_a(t)$) creating the interference. With a sufficiently large number m , this average sample value group - consisting of n average signal sampling values - represents a period of the signal components, i.e. in particular the additive static hum components, coupled in phase and frequency with the signal ($f_a(t)$) creating the interference and contained in the useful signal ($e_a(t)$) suffering from interference. An analogue compensation signal ($k_a(t)$) results, after digital to analogue conversion, from a periodical

repetition of the averaged sample value group which is phase locked to the signal ($f_a(t)$) creating the interference. This repetition may occur e.g. by further circulation of the sampling values accumulated in the shift register (6). A subtraction of this compensation signal ($k_a(t)$) from the useful signal ($e_a(t)$) suffering from interference yields the desired, freed from interference or having reduced interference, output signal ($a_a(t)$).

Fig. 2 shows the basic circuit diagram of a circuit for digital input and output signals operating in accordance with the same principle. All the signals are correspondingly indexed with a "d". Instead of the low pass filter (1) and the A/D converter (2) of Fig. 1, for a useful digital signal ($e_d(t)$) suffering from interference, a digital low pass filter (11) and a sampling rate converter (12) are utilised which converts the sampling rate of e.g. 44.1 kHz of a digital signal of a compact disc to the nth multiple of the frequency of the signal ($f(t)$) creating the interference. The switch (10) in Fig. 1 is replaced by switches (27). The number of switches (27) corresponds to the bit length of a sampling value. The conversion in the reverse direction then takes place in a sampling rate converter (18). The resulting correcting value ($k_d(t)$) is subtracted from the useful digital signal ($e_d(t)$) suffering from interference in a subtracting circuit (19). The other circuit components correspond to those in Fig. 1.

By means of the compensating circuits depicted in Fig. 1 and Fig. 2, there is realised with simple digital means, a frequency and phase selective filter whose selection frequency is determined, inter alia, by the signal ($f(t)$) creating the interference.



Claims

1. Method for the digital compensation of a predetermined interference signal ($n(t)$) which arises by virtue of stray electrical or magnetic pick-up caused by an interference signal ($f(t)$) and which occurs in a useful signal ($e(t)$) suffering from interference through additive mixing with a useful signal ($s(t)$), characterised in that, a group of n sampling values is extracted from the useful signal suffering from interference during each of a total of m periods (m an integer) of the signal ($f(t)$) creating the interference and at the clock pulse rate (T) of an n th multiple frequency (n an integer) generated in phase with the signal ($f(t)$) creating the interference, that an average sampling group having n sampling values is determined from these m groups of sampling values having respectively n sampling values by arithmetical averaging of the sampling values extracted with the same respective phase relationship to the signal ($f(t)$) creating the interference, that this average sampling group coupled in phase with the signal ($f(t)$) creating the interference is cyclically repeated with the original phase relationship, and that a thus formed compensating signal ($k(t)$) is continually subtracted from the useful signal ($e(t)$) suffering from interference.
2. Method in accordance with Claim 1, characterised in that, the clock pulse rate (T) is obtained from the voltage of the local energy supply mains by phase coupled frequency multiplication if a mains induced hum component is to be removed.
3. Method in accordance with Claim 1, characterised in that, the useful signal ($e(t)$) suffering from interference is filtered in an analogue or digital low pass filter (1) before the sampling, that the low pass frequency is selected such that an aliasing interference is not generated by the succeeding sampling, and that phase rotations of the spectral components of the interference signal ($n(t)$) are avoided or compensated by the filtering.
4. Method in accordance with Claim 1, characterised in that, the periodic number m of the m groups of sampling values used for the averaging is a power of 2.
5. Method in accordance with Claim 1, characterised in that, the m groups of sampling values are determined from the useful signal ($e(t)$) suffering from interference during m contiguous periods of the signal ($f(t)$) creating the interference.



6. Method in accordance with Claim 1, characterised in that, the m groups of sampling values are determined from the useful signal $(e(t))$ suffering from interference during m non-contiguous periods of the signal $(f(t))$ creating the interference.

7. Method in accordance with any one or more of the preceding Claims, characterised in that, the time interval between each two of the m sampling periods amounts to a fixed integral multiple of the periodic time of the signal $(f(t))$ creating the interference.

8. Method in accordance with any one or more of the preceding Claims, characterised in that, the time interval between each two of the m sampling periods is determined by a random number generator and amounts to an integral multiple of the periodic time of the signal $(f(t))$ creating the interference.

9. Method in accordance with any one or more of the preceding Claims, characterised in that, the useful signal $(e(t))$ suffering from interference is sampled and averaged by the formation of a moving average during each m periods of the signal $(f(t))$ creating the interference and the result of the respective last averaging is utilised in stages as compensating signal $(k(t))$ during each individual period of the signal $(f(t))$ creating the interference.