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(54) **Audio processor for converting a mono signal to a stereo signal**

(57) An audio processor (or mono-to-stereo converter) arranged to receive a single channel audio input signal X and generate a set of stereo audio output signals L, R in response. The outputs L, R are based on four delayed versions S1, S2, S3, S4 of the input signal X. S1 is delayed by delay d1 in relation to X, and S2 is delayed by delay d2 in relation to S1. S3 is delayed by a delay d3 in relation to X, and S4 is delayed by delay d4 in relation to S3. The output L is then generated as a sum of X, S1, and S4, while the output R is generated as a sum of X, S2, and S3. Delays d1 and d3 are selected to be different

and within a range of 20 ms to 100 ms, e.g. d1=50 ms and d3 =60 ms. Delays d2 and d4 are selected to be within 50 μ s to 1 ms, e.g. 450-650 μ s. Such processor produces a stereo signal suited for headphone listening without the feeling of in-head localization and still with a natural timbre. Additionally, low-pass or band-pass filters and appropriate gains can be applied for further refinement. The audio processor can be implemented with a low signal processing requirement and is thus suited as mono-to-stereo converter in portable equipment such as mobile phones, hearing aids etc.

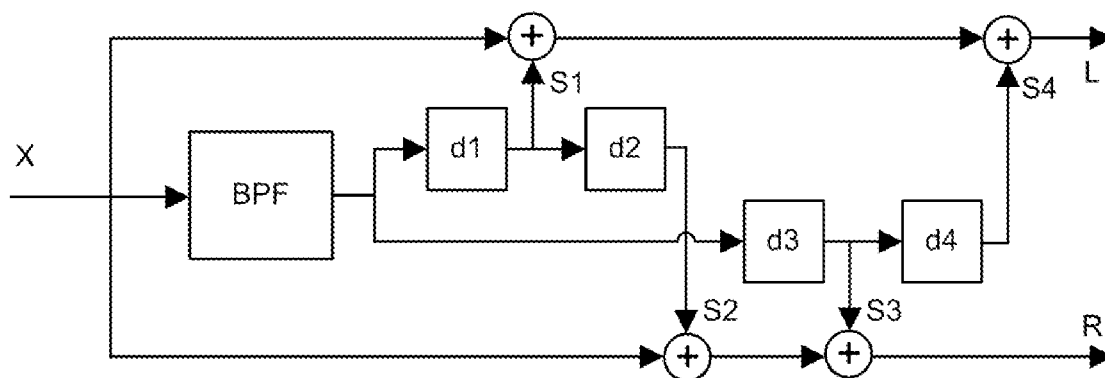


Fig. 1

Description

FIELD OF THE INVENTION

[0001] The present invention relates to the field of signal processing, especially audio signal processing. More specifically, the invention provides an audio processor capable of converting an audio input signal (mono) to two audio output signals (stereo).

BACKGROUND OF THE INVENTION

[0002] When listening to a monophonic signal through stereo headphones the sound image appears to be inside (in the middle) of the head. This is referred to as in-the-head-localization. This is undesirable and is generally experienced to be unpleasant and unnatural since it does not occur in real-life listening where the human hearing will normally always be able to appoint a position in space to a sound source. Furthermore, it can lead to listener fatigue when exposed for longer periods of time. It is therefore desirable to enhance this impoverished sound image in such a way that it is more natural and pleasant to listen to. The enhancement should be applicable not only to a single voice or musical instrument, but to the final mix of signals. The processing should be done without using excessive signal processing power, since implementations typically have to be portable.

[0003] US 6,084,970 describes a method and a device for converting a monophonic signal into a stereo signal by selectively allocating frequency bands of the input signal to left or right outputs. In this way, some frequency components will be present only in the left output and others only in the right output. This method can be used successfully for processing a single audio track when mixing it into a final mix, especially if the mix already contains some stereo content. This signal should then be played through loudspeakers. However, when presented through headphones, this can lead to an unpleasant sound - especially when this effect is applied to a final mix. Furthermore, the method is not able to provide out-of-head localization. Another disadvantage of the method is that a reasonably large number of filters are needed which leads to an unacceptably high signal processing requirement.

[0004] A way to alleviate the problem of in-the-head-localization is by means of binaural synthesis, such as described in J. Blauert, "Spatial hearing: The psychophysics of human sound localization", MIT Press, Cambridge, USA, Revised edition, 1997. By employing head-related transfer functions (HRTFs) a sound can be processed such that it appears to be placed at some location outside the head. It is however, well known that placing sound sources convincingly in front of the listener is exceedingly difficult. The location of the sound source is often confused with other positions in the median plane and very often in-the-head-localization occurs. It is especially difficult to simulate distance when using HRTFs

alone, and furthermore, the sound colour, i.e. the timbre, of the signal can be affected adversely.

[0005] Another way to convert a monophonic signal to a stereo signal is by applying traditional stereo reverberation processing such as described in M. Kahrs, K. Brandenburg, "Applications of digital signal processing", Kluwer Academic Publishers, Boston, USA, 1998. This is commonly used in professional mixing studios. Applying reverb to a "dry" voice of musical instrument signal is well-known to give a more pleasant sound. However, reverberation is generally not applied to the final mix, as this typically leads to blurring of the sound image. When listening through headphones, applying traditional stereo reverberation to a monophonic signal does not create out-of-head localization. Furthermore, the implementation of traditional stereo reverberation requires a very large amount of processing power.

SUMMARY OF THE INVENTION

[0006] Thus, according to the above description, it is an object of the present invention to provide a simple audio processor capable of converting a mono signal into a stereo signal which is suited for headphone listening without in-head localization, and wherein the stereo signal does not suffer from severe timbre distortion.

[0007] In a first aspect, the invention provides an audio processor arranged to receive an audio input signal and generate first and second audio output signals in response thereto, the processor being arranged to

- generate a first delayed version of the audio input signal being delayed by a first delay in relation to the audio input signal,
- generate a second delayed version of the audio input signal, delayed by a second delay in relation to the first delayed version of the audio input signal,
- generate a third delayed version of the audio input signal being delayed by a third delay in relation to the audio input signal,
- generate a fourth delayed version of the audio input signal, delayed by a fourth delay in relation to the third delayed version of the audio input signal,
- generate the first audio output signal as a first sum of the audio input signal, the first and the fourth delayed versions of the audio input signal,
- generate the second audio output signal as a second sum of the audio input signal, the second and the third delayed versions of the audio input signal, wherein the first and third delays are within a first delay range of 20 ms to 100 ms, the first and third delays being different,

and wherein the second and fourth delays are within a second delay range of 50 μ s to 1 ms.

[0008] Such an audio processor is capable of taking a single channel audio input signal and provide a stereo output signal in response, i.e. mono-to-stereo conversion

and thus provides what could be called "mono-widening". The stereo output signal is suited for stereo headphone listening since it provides the listener with an out-of-head localization without severely changing the timbre of the single channel input signal. The audio processor can be seen as providing a widening effect by simulating a very simple virtual sound source in front of the listener with a single reflection on each side of the listener, each of these reflections being provided with an ipsi-lateral and a contra-lateral contribution. With very little processing power required, the audio processor according to the first aspect is capable of providing an out-of-head localization without suffering from severe coloration, when listened to over headphones.

[0009] The mono-to-stereo conversion is obtained with very simple processing means in the form of delays and summations and optionally first order filters. Thus, it can be implemented in miniature portable devices with very limited processing capacity, such as mobile phones, hearing aids etc. In contrast to prior art mono-to-stereo converters, the audio processor according to the first aspect provides a stereo output signal which is based on the unprocessed input signal, and this helps to ensure that the stereo output signal will have the general timbre in common with the input signal, whereas the delayed versions of the input signal serve to simulate simple acoustic reflections which help to provide an out-of-head localization.

[0010] In a preferred embodiment, the first audio output signal consists only of a sum of the audio input signal and the first and fourth delayed versions of the audio input signal, and the second audio output signal consists only of a sum of the audio input signal and the second and third delayed versions of the audio input signal, such as defined above. This embodiment is very simple and can be implemented with very limited processing required, and still with a natural timbre and out-of-head localization. However, an improved sound quality can be obtained e.g. by adding more delayed versions of the audio input signal to each of the two output signals, providing appropriate filtering and attenuations, as will be explained in the following.

[0011] The first delay range is preferably 30 ms to 80 ms, such as 40 ms to 70 ms, such as 50 ms to 60 ms. Such first delay ranges are found suitable for simulating simplified lateral reflections that result in out-of-head localization.

[0012] The second delay range is preferably 50 μ s to 800 μ s, 200 μ s to 700 μ s, such as 450 μ s to 650 μ s. Such second delay ranges are found suitable for supporting simplified contra-lateral versions of the reflections, since the second delay range is thus within interaural time differences experienced in real-life listening.

[0013] The first and third delays are preferably selected such that a difference between the first delay and the third delay is within a third relay range of 1 ms to 30 ms, such as 3 ms to 15 ms, such as 5 ms to 10 ms. Such delay difference is found suitable for simulating the effect

of asymmetric reflections. Compared with identical first and third delays, this delay difference supports out-of-head localization.

[0014] The audio processor may include a low-pass filter section arranged to

- low-pass filter the first delayed version of the audio input signal before being summed to form the first audio output signal, and
- low-pass filter the third delayed version of the audio input signal before being summed to form the second audio output signal.

[0015] Such a low-pass filter section may further be arranged to

- low-pass filter the second delayed version of the audio input signal before being summed to form the first audio output signal, and
- low-pass filter the fourth delayed version of the audio input signal before being summed to form the second audio output signal.

[0016] Low-pass filtering of the delayed versions of the audio input signal before adding these versions to the audio input signal generally provides a more natural sounding output signal, since the timbre at high frequencies is less influenced by the delayed versions of the audio input signal.

[0017] The low-pass filter section may be arranged to provide a cut-off frequency within the range 300 Hz to 5 kHz, such as within the range 400 Hz to 3 kHz, such as 500 Hz to 1 kHz.

[0018] The audio processor may include a band-pass filter section arranged to band-pass filter all of the first, second, third and fourth delayed versions of the audio input signal before being summed to form the first and second audio output signals. By band-pass filtering the delayed versions of the audio input signal, it is ensured that the unprocessed audio input signal serves to determine the timbre both at low and high frequencies, whereas the delayed signals only provide a major contribution in the mid-frequency range. In a special implementation, all of the first, second, third and fourth delayed version of the audio input signal are based on a band-pass filtered version of the audio input.

[0019] The band-pass filter section may be arranged to provide a band-pass frequency range of 100 Hz to 5 kHz, such as 300 Hz to 3 kHz, such as 500 Hz to 1 kHz.

[0020] All filters, low-pass filters or band-pass filters, may be implemented with first order filter sections, which are easy to implement and still provides the intended effect without the complexity required by higher order filters.

[0021] The first sum may include a fifth delayed version of the audio input signal, and wherein the second sum includes a sixth delayed version of the audio input signal. As mentioned, further delayed versions may be added,

also more than the mentioned fifth and sixth delayed versions. In general, the first and second sum may further include a plurality of delayed versions of the audio input signal. Each of this plurality of delayed versions of the audio input signal is preferably provided with different delays so as to simulate multiple reflections. Such higher number of delayed versions of the audio input signal can be used for further increasing sound quality, but at the price of required processing power.

[0022] The audio processor may include an attenuation section arranged to attenuate the first, second, third and fourth delayed versions in relation to the audio input signal before generating the first and second sums. In preferred embodiments, the delayed versions are attenuated so as to further increase the influence of the audio input signal itself in the output signals, thus resulting in a more natural timbre. The first, second, third and fourth delayed versions may be attenuated by at least 2 dB in relation to the audio input signal, such as by at least 4 dB, such as by at least 6 dB, such as by at least 10 dB.

[0023] The audio processor may be arranged to attenuate the audio input signal prior to generating the delayed versions of the audio input signal. Especially such attenuation may be used to provide an unchanged loudness when switching from unprocessed (i.e. providing the audio input signal in both output channels) to processed (outputting the first and second audio output signals). Such constant loudness may typically be obtained by an attenuating the audio input signal by 2-3 dB prior to generating the delayed versions of the audio input signal.

[0024] A second aspect of the invention provides a device including an audio processor according to the first aspect. As mentioned, the audio processor can advantageously be built into portable devices for converting mono audio signals to stereo audio signals which are more pleasant for stereo playback.

[0025] A non-exhaustive list of such types of devices is: mobile phones, portable computers, headphones, headsets, assistive listening devices, hearing aids and a set of loudspeakers arranged for positioning close to a listener's ears.

[0026] The device may also be in the form of a stand-alone mono-to-stereo converter device arranged for wired or wireless receipt of the (mono) audio input signal and for wired or wireless output of the (stereo) first and second audio output signals.

[0027] A third aspect of the invention provides a system including

- a device according to the second aspect, i.e. a device including an audio processor according to the first aspect, and
- first and second electro-acoustic transducers arranged to
 - convert the respective first and second audio output signals into according respective first and second acoustic signals, and

- to playback the respective first and second acoustic signals to the respective left and right ears of a user.

[0028] The first and second electro-acoustic transducers may be included in one of: a set of headphones, a headset, a set of hearing aids, a set of hearing assistive devices, a set of loudspeakers arranged for positioning close to a listener's ears.

[0029] The system may include a set of left and right hearing aid devices arranged for position in respective left and right ears of a user. The invention is advantageous even for hearing aids which typically has very limited processing power available due to the small space combined with requirements for extremely low power consumption. Thus, using the invention for converting a mono signal from e.g. a CD player, an MP3 player or the like, it is possible to provide a pleasant sound reproduction even though the available bandwidth only allows transmission of a mono signal to the hearing aids.

[0030] In one such hearing aid embodiment, the audio processor is included in a separate unit, wherein the separate unit is arranged to receive the audio input signal from an external device, and to provide the first and second audio output signals to the respective left and right hearing aid devices. In this embodiment, the mono-to-stereo conversion is performed by the separate unit and not in the hearing aid devices, thus saving processing power in the hearing aid devices. The separate unit may include a wireless transmitter arranged to transmit signals representing the first and second audio output signals to respective receivers in the left and right hearing aid devices. Alternatively, the separate unit has a wired connection to the left and right hearing aid devices.

[0031] Another such hearing aid embodiment includes a separate unit arranged to receive the audio input signal from an external device, and to provide this audio input signal to both of the left and right hearing aid devices, wherein the left hearing aid device includes a processor arranged to generate the first audio output signal as a first sum of the audio input signal and first and second delayed versions of the audio input signal, and the right hearing aid device includes a processor arranged to generate the second audio output signal as a second sum of the audio input signal and third and fourth delayed versions of the audio input signal. The audio processor according to the invention is suited for splitting into two separate parts which can be implemented in respective left and right hearing aid devices. The separate unit may include a wireless transmitter arranged to transmit a signal representing the audio input signal to respective receivers in the left and right hearing aid devices. Alternatively, the separate unit has a wired connection to the left and right hearing aid devices.

[0032] In a fourth aspect, the invention provides a method for converting a single audio input signal (X) to a set of first and second audio output signals, the method including

- generating a first delayed version of the audio input signal being delayed by a first delay in relation to the audio input signal,
- generating a second delayed version of the audio input signal, delayed by a second delay in relation to the first delayed version of the audio input signal,
- generating a third delayed version of the audio input signal being delayed by a third delay in relation to the audio input signal,
- generating a fourth delayed version of the audio input signal, delayed by a fourth delay in relation to the third delayed version of the audio input signal,
- generating the first audio output signal as a first sum of the audio input signal, the first and the fourth delayed versions of the audio input signal,
- generating the second audio output signal as a second sum of the audio input signal, the second and the third delayed versions of the audio input signal, wherein the first and third delays are within a first delay range of 20 ms to 100 ms, the first and third delays being different,

and wherein the second and fourth delays are within a second delay range of 50 μ s to 1 ms.

[0033] In a fifth aspect, the invention provides computer-executable program code arranged to perform the method according to the fourth aspect. The program code may be dedicated program code for a specific signal processor, or program code arranged for a general purpose computer, e.g. a Personal Computer.

[0034] In a sixth aspect, the invention provides a data carrier including a computer executable program code according to the fifth aspect. The data carrier may be such as any type of disk, memory card, memory stick, hard disk etc.

[0035] It is appreciated that the same advantages and embodiments described for the first aspect apply as well for the second, third, fourth, fifth and sixth aspects. Further, it is appreciated that the described embodiments can be intermixed in any way between all the mentioned aspects.

BRIEF DESCRIPTION OF THE FIGURES

[0036] The invention will now be described in more detail with regard to the accompanying figures of which

Fig. 1 illustrates a diagram of a simple embodiment,

Fig. 2 illustrates a diagram of another simple embodiment,

Fig. 3 illustrates a diagram of yet another embodiment,

Fig. 4 illustrates a diagram of an embodiment split into two separate algorithms - one for each audio output,

Fig. 5 illustrates a diagram of an embodiment with several gains, delays and filters,

Fig. 6 illustrates a hearing aid system embodiment with mono-to-stereo conversion in a separate unit, and

Fig. 7 illustrates a hearing aid system embodiment with wireless transmission of a mono signal to separate left and right audio processor parts in each hearing aid.

[0037] The figures illustrate specific ways of implementing the present invention and are not to be construed as being limiting to other possible embodiments falling within the scope of the attached claim set.

DETAILED DESCRIPTION OF EMBODIMENTS

[0038] Fig. 1 shows a signal diagram with basic elements of a simple mono-to-stereo algorithm embodiment for use in a mono-to-stereo audio processor. Four delays d1, d2, d3, d4 are used, namely delays corresponding to the respective first, second, third and fourth delays as defined in the preceding chapter.

[0039] Following the description in the preceding chapter, the respective first and second sums are each seen to be split into two summation points in this specific embodiment. As seen, the mono audio input signal X is directly applied to left L and right R output signals via summation two points, and thus the unprocessed audio input signal X forms a vital part of both of the left L and right R output signals.

[0040] An optional band-pass filter BPF serves to provide a band-pass filtered version of the audio input signal X. This band-pass filtered version of the input signal X is then used as input for providing band-pass filtered delayed versions S1, S2, S3, S4 of the input signal X. The band-pass filter BPF can in principle be omitted completely, or replaced by only a low-pass filter or a high-pass filter. However, with the band-pass filter BPF included, it is ensured that delayed versions of the lowest and highest audio frequencies are not passed to the output signals L, R. Hereby, the widening effect provided by these delayed versions of the input signal X only causes a minimal influence on the overall perceived timbre when compared to the timbre of the unprocessed input signal X.

[0041] By the use of two summation points for each of the outputs L, R, it is ensured that the left output signal L is a sum of 1) the input signal X, 2) signal S1 being a delayed version of the input signal X, delayed by delay d1, and 3) signal S4 also being a delayed version of the input signal X, delayed by delay d3 and further delayed by delay d4. The right output signal R is sum of 1) the input signal X, 2) signal S3 being a delayed version of the input signal X, delayed by delay d3, and 3) signal S2 also being a delayed version of the input signal X, namely S1 delayed by delay d1 and further delayed by delay d2.

[0042] The signal diagram illustrated in Fig. 1 is simple and thus suited for implementation on processors with limited signal processing capabilities. If the required memory is available, the delays d1, d2, d3, d4 can preferably be implemented purely as simple sample buffers which would further help to save processing power compared to alternative delay implementations.

[0043] In a specific embodiment, the band-pass filter BPF and delays d1, d2, d3, d4 of Fig. 1 can be chosen as follows: BPF with both high and low frequency cut-off at 500 Hz and 1000 Hz respectively, d1=50 ms, d2=650 μ s, d3=60 ms, d4=650 μ s. Thus, d1 and d3 are chosen to have a difference of 10 ms which is found suitable to provide a widening effect and thus avoid in-head localization. Delays d2 and d4 may likewise be chosen slightly different, but this is not essential.

[0044] More delayed version of the input signal X can be added to form the output signals L, R, but a preferred embodiment only consists of two delayed versions of the input signal X per output signal L, R, such as illustrated in Fig. 1, thereby providing a very simple algorithm.

[0045] Fig. 2 illustrates a signal diagram of another simple embodiment serving to illustrate the core elements of the invention. Here only two summation points are used for providing the left and right outputs L, R based on the input signal X and delayed versions S1, S2, S3, S4 thereof. Delays d1 and d3 for providing delayed versions S1, S3 of the input signal X are similar to those of Fig. 1. However, delays d12, d14 for providing respective delayed versions S2, S4 of the input signal X are different from delays d2, d4 of Fig. 1. In Fig. 2 input signal X is used as input to delays d12, d14. Thus, d12 should be selected as the sum of d1 and d2 in Fig. 1, and accordingly d14 should be selected as the sum of d3 and d4 to provide similar output signals L, R. As in Fig. 1, an optional band-pass filter BPF is included so that the delayed versions S1, S2, S3, S4 are band-pass filtered versions of the input signal X. A simple way of reducing the timbre impact of these signals S1, S2, S3, S4 on the output signals L, R, is to gain down these signals S1, S2, S3, S4 in relation to the input signal X, before being added to the input signal X, e.g. by providing a gain of -3 dB, -6 dB or -10 dB, as will be illustrated in the following.

[0046] Fig. 3 illustrates a signal diagram of another embodiment where the delayed versions of the input signal X are provided with a different layout of delays D1, D2, D3, D4 to provide a set of output signals L, R in response to the input signal X. Thus, note that these delays D1, D2, D3, D4 are different from delays d1, d2, d3, d4 of Fig. 1, and thus should be selected differently to provide the same result. In Fig. 3, a first gain G1, e.g. -2 dB, is included to attenuate the input signal X prior to being used for the further processing. Hereby the processor can be switched in and out without altering the overall perceived loudness. A second gain G2, e.g. of -6 dB, and a band-pass filter BPF is included to generally attenuate and band-pass filter all delayed versions of the input signal X.

[0047] Fig. 4 illustrates a signal diagram of an audio processor embodiment split into two separate parts, namely one taking the input signal X and converts it to a first output signal L, and one taking the input signal X and converts it to a second output signal R. Thus, this embodiment is suited e.g. for hearing aids where each of the separate parts can be implemented into respective left and right hearing aid devices. It is appreciated that delays D5, D6, D7, D8 are different from the delays in the earlier Figures, and thus should be selected so as to obey what is generally described for Fig. 1. Respective gains and band-pass filters G3, BPF and G5, BPF are included so that all delayed versions of the input signal X are gained and band-pass filtered prior to being summed. Further, gains G4, G6 are included to gain the respective summed signals prior to being output as respective first and second output signals L, R. By proper selection of gains G4, G6, the audio processor can be switched in and out without altering the overall perceived loudness. Thus, it becomes possible for the user to switch between "mono" and "stereo" without any appreciable change in loudness.

[0048] Fig. 5 illustrates signal diagram of a more complex embodiment. Delays D9, D10, D11, D12 are used for generating delayed versions of the input signal X, as explained earlier, prior to being summed to form respective output signals L, R. Further, gains G9, G10, G11, G12 are included to allow an individual attenuation of the contributions to the output signals L, R. This allows the choice of a higher attenuation for the contra-lateral contributions compared to the ipsi-lateral contributions, which is considered as advantageous, e.g. G9 and G11 can be chosen as -6dB, whereas G10 and G12 can be chosen as -9dB. In order to allow the processor to be switched in and out without altering the overall perceived loudness, G7 and G8 can be chosen as -2dB.

[0049] As seen, the input signal X is filtered by filters Filter1, Filter4 prior to being fed to the summation points where it is summed with delayed versions of the input signal X. The delayed versions of the input signal X are based on filtered versions of the input signal X filtered by filters Filter2, Filter3. All of Filter1, Filter2, Filter3, Filter4 can be chosen to be different or be chosen to be similar, and they can be chosen to be low-pass, high-pass or band-pass filters or more complex filters. In more complex embodiments suited for applications with more processing power available, the filters Filter1, Filter2, Filter3, Filter4 can be chosen to be HRTFs representing a desired 3D direction, thus giving the listener a more precise experience of localizing a virtual sound source. Specifically, the filters Filter1 and Filter4 may implement the HRTFs in front of the listener, while the filter Filter2 implements an ipsi-lateral HRTF for a direction on the side of the head, whereas the filter Filter3 implements the corresponding contra-lateral HRTF for the same direction on the side of the head.

[0050] Fig. 6 illustrates a hearing aid system in schematic. Hearing aid devices HL, HR are suited for a lis-

tener's respective left and right ear. The hearing aid devices HL, HR have respective electronic circuits EL, ER connected to drive miniature loudspeakers, called receivers in hearing aids. In one mode of operation, the electronic circuits EL, ER are connected to microphones in order to amplify sound captures in the environments. In another mode of operation, the hearing aid system can be connected to an external sound source that provides an audio input signal X, either an electrical wired input signal or an input signal in a wireless form. The input signal X is received by a processor unit PU that includes an audio processor AP, such as explained above. This audio processor AP generates a set of stereo output signals L, R based on the single channels input X. This set of stereo output signals L, R are then applied to the respective electronic circuits EL, ER of the hearing aid devices HL, HR. Hereby, the listener wearing the hearing aid devices HL, HR can listen to the acoustic left and right output signals AL, AR based on sound from a connected external sound source (such as a CD player, an MP3 player, a radio or a mobile phone etc.) in a pleasant way, even though the original sound from the external sound source is in the form of a mono signal. In this embodiment the audio processor AP converting the mono signal X to a stereo signal L, R is included in a separate unit PU that can be applied with a suitable signal processor, still with a size to fit in a user's pocket. The stereo signals L, R can be applied to the hearing aid devices HL, HR by wire or in wireless form.

[0051] Fig. 7 illustrates another hearing aid system embodiment capable of generating sound based on an input signal X from an external sound source. Here, the processor unit PU receiving the input signal X does not convert the single channel input X to a stereo signal, but transmit the single channel input signal X to both of the hearing aid devices HL, HR in the form of a Radio Frequency (RF) signal by means of a built-in RF transmitter RFT. Each of the hearing aid devices HL, HR has RF receivers RFL, RFR arranged to receive the RF signal transmitted from the processor unit PU. The input signal X can then be regenerated based on the received RF signals. Each hearing aid device HL, HR then has one part of an audio processor APL, APR, e.g. such as illustrated and explained in connection with Fig. 4. These separate audio processor parts APL, APR then generate different left L and right R output signals that are applied to miniature loudspeakers that generate the acoustic signals AL, AR in response. In this embodiment, the mono-to-stereo processing is implemented in processors of the hearing aid devices HL, HR. However, as mentioned, the simple nature of the audio processor and method for converting a mono signal to a stereo signal according to the invention, it is possible to implement this processing in existing hearing aid processors.

[0052] It is appreciated that the principles illustrates in Fig. 6 and 7 may alternatively be implemented in a stereo headphone, a headset, an assistive listening device, hearing aids, a set of loudspeakers arranged for posi-

tioning close to a listener's ears etc.

[0053] To sum up: the invention provides an audio processor (or mono-to-stereo converter) arranged to receive a single channel audio input signal X and generate a set of stereo audio output signals L, R in response. The outputs L, R are based on four delayed versions S1, S2, S3, S4 of the input signal X. S1 is delayed by delay d1 in relation to X, and S2 is delayed by delay d2 in relation to S1. S3 is delayed by a delay d3 in relation to X, and S4 is delayed by delay d4 in relation to S3. The output L is then generated as a sum of X, S1, and S4, while the output R is generated as a sum of X, S2, and S3. Delays d1 and d3 are selected to be different and within a range of 20 ms to 100 ms, e.g. d1=50 ms and d3=60 ms. Delays d2 and d4 are selected to be within 50 μ s to 1 ms, e.g. 450-650 μ s. Such processor produces a stereo signal suited for headphone listening without the feeling of in-head localization and still with a natural timbre. Additionally, low-pass or band-pass filters and appropriate gains can be applied for further refinement. The audio processor can be implemented with a low signal processing requirement and is thus suited as mono-to-stereo converter in portable equipment such as mobile phones, hearing aids etc.

[0054] Although the present invention has been described in connection with the specified embodiments, it should not be construed as being in any way limited to the presented examples. The scope of the present invention is to be interpreted in the light of the accompanying claim set. In the context of the claims, the terms "including" or "includes" do not exclude other possible elements or steps. Also, the mentioning of references such as "a" or "an" etc. should not be construed as excluding a plurality. The use of reference signs in the claims with respect to elements indicated in the figures shall also not be construed as limiting the scope of the invention. Furthermore, individual features mentioned in different claims, may possibly be advantageously combined, and the mentioning of these features in different claims does not exclude that a combination of features is not possible and advantageous.

Claims

1. An audio processor arranged to receive an audio input signal (X) and generate first and second audio output signals (L, R) in response thereto, the processor being arranged to
 - generate a first (S1) delayed version of the audio input signal (X) being delayed by a first (d1) delay in relation to the audio input signal (X),
 - generate a second (S2) delayed version of the audio input signal (X), delayed by a second delay (d2) in relation to the first (S1) delayed version of the audio input signal (X),
 - generate a third (S3) delayed version of the

audio input signal (X) being delayed by a third (d3) delay in relation to the audio input signal (X),
 - generate a fourth (S4) delayed version of the audio input signal (X), delayed by a fourth delay (d4) in relation to the third (S3) delayed version of the audio input signal (X),
 - generate the first audio output signal (L) as a first sum of the audio input signal (X), the first (S1) and the fourth (S4) delayed versions of the audio input signal (X),
 - generate the second audio output signal (R) as a second sum of the audio input signal (X), the second (S2) and the third (S3) delayed versions of the audio input signal (X),

wherein the first (d1) and third (d3) delays are within a first delay range of 20 ms to 100 ms, the first (d1) and third (d3) delays being different, and wherein the second (d2) and fourth (d4) delays are within a second delay range of 50 μ s to 1 ms.

2. Audio processor according to claim 1, wherein the first delay range is 30 ms to 80 ms, such as 40 ms to 70 ms, such as 50 ms to 60 ms.
3. Audio processor according to claim 1 or 2, wherein the second delay range is 50 μ s to 800 μ s, 200 μ s to 700 μ s, such as 450 μ s to 650 μ s.
4. Audio processor according to any of the preceding claims, wherein the first and third delays (d1, d3) are selected such that a difference between the first delay (d1) and the third delay (d3) is within a third delay range of 1 ms to 30 ms, such as 3 ms to 15 ms, such as 5 ms to 10 ms.
5. Audio processor according to any of the preceding claims, including a low-pass filter section arranged to
 - low-pass filter the first (S1) delayed version of the audio input signal (X) before being summed to form the first audio output signal (L), and
 - low-pass filter the third (S3) delayed version of the audio input signal (X) before being summed to form the second audio output signal (R).
6. Audio processor according to claim 5, wherein the low-pass filter section is arranged to
 - low-pass filter the second (S2) delayed version of the audio input signal (X) before being summed to form the first audio output signal (L), and
 - low-pass filter the fourth (S4) delayed version of the audio input signal (X) before being summed to form the second audio output signal (R).

7. Audio processor according to claim 5 or 6, wherein the low-pass filter section is arranged to provide a cut-off frequency within the range 300 Hz to 5 kHz, such as within the range 400 Hz to 3 kHz, such as 500 Hz to 1 kHz.
8. Audio processor according to any of claims 5-7, including a band-pass filter section arranged to band-pass filter all of the first, second, third and fourth (S1, S2, S3, S4) delayed versions of the audio input signal (X) before being summed to form the first and second audio output signals (L, R).
9. Audio processor according to claim 8, wherein the band-pass filter section is arranged to provide a band-pass frequency range of 100 Hz to 5 kHz, such as 300 Hz to 3 kHz, such as 500 Hz to 1 kHz.
10. Audio processor according to any of claims 5 to 9, wherein all filter sections are implemented as first order filters.
11. Audio processor according to any of the preceding claims, wherein the first sum includes a fifth (S5) delayed version of the audio input signal (X), and wherein the second sum includes a sixth (S6) delayed version of the audio input signal (X).
12. Audio processor according to any of the preceding claims, including an attenuation section arranged to attenuate the first, second, third and fourth delayed versions (S1, S2, S3, S4) in relation to the audio input signal (X) before generating the first and second sums.
13. Audio processor according to claim 12, wherein the first, second, third and fourth delayed versions (S1, S2, S3, S4) of the audio input signal (X) are attenuated by at least 2 dB in relation to the audio input signal (X), such as by at least 4 dB, such as by at least 6 dB, such as by at least 10 dB.
14. Audio processor according to any of the preceding claims, arranged to attenuate the audio input signal (X) prior to generating the delayed versions (S1, S2, S3, S4) of the audio input signal (X).
15. Device (PU) including an audio processor (AP) according to any of claims 1-14.
16. System including
 - a device (PU) according to claim 15, and
 - first and second electro-acoustic transducers arranged to
 - convert the respective first and second audio output signals (L, R) into respective first and second acoustic signals (AL, AR), and

- to playback the respective first and second acoustic signals (AL, AR) to the respective left and right ears of a user.
17. System according to claim 16, wherein the first and second electro-acoustic transducers are included in one of: a set of headphones, a headset, a set of hearing aids (HL, HR), a set of hearing assistive devices, a set of loudspeakers arranged for positioning close to a listener's ears.
18. System according to claim 16 or 17, including a set of left and right hearing aid devices (HL, HR) arranged for position in respective left and right ears of a user.
19. System according to claim 18, including a separate unit (PU) including the audio processor (AP), wherein the separate unit (PU) is arranged to receive the audio input signal (X) from an external device, and to provide the first and second audio output signals (L, R) to the respective left and right hearing aid devices (HL, HR).
20. System according to claim 19, wherein the separate unit (PU) includes a wireless transmitter arranged to transmit signals representing the first and second audio output signals to respective receivers in the left and right hearing aid devices.
21. System according to claim 18, including a separate unit (PU) arranged to receive the audio input signal (X) from an external device, and to provide this audio input signal (X) to both of the left and right hearing aid devices, wherein the left hearing aid device (HL) includes a processor (APL) arranged to generate the first audio output signal (L), and the right hearing aid device (HR) includes a processor (APR) arranged to generate the second audio output signal (R).
22. System according to claim 21, wherein the separate unit (PU) includes a wireless transmitter (RFT) arranged to transmit a signal representing the audio input signal (X) to respective receivers (RFL, RFR) in the left and right hearing aid devices (HL, HR).
23. Method for converting a single audio input signal (X) to a set of first and second audio output signals (L, R), the method including
- generating a first (S1) delayed version of the audio input signal (X) being delayed by a first (d1) delay in relation to the audio input signal (X),
 - generating a second (S2) delayed version of the audio input signal (X), delayed by a second delay (d2) in relation to the first (S1) delayed version of the audio input signal (X),
 - generating a third (S3) delayed version of the audio input signal (X) being delayed by a third (d3) delay in relation to the audio input signal (X),
 - generating a fourth (S4) delayed version of the audio input signal (X), delayed by a fourth delay (d4) in relation to the third (S3) delayed version of the audio input signal (X),
 - generating the first audio output signal (L) as a first sum of the audio input signal (X), the first (S1) and the fourth (S4) delayed versions of the audio input signal (X),
 - generating the second audio output signal (R) as a second sum of the audio input signal (X), the second (S2) and the third (S3) delayed versions of the audio input signal (X),
- wherein the first (d1) and third (d3) delays are within a first delay range of 20 ms to 100 ms, the first (d1) and third (d3) delays being different, and wherein the second (d2) and fourth (d4) delays are within a second delay range of 50 μ s to 1 ms.
24. Computer executable program code arranged to perform the method according to claim 23.
25. Data carrier including a computer executable program code according to claim 24.

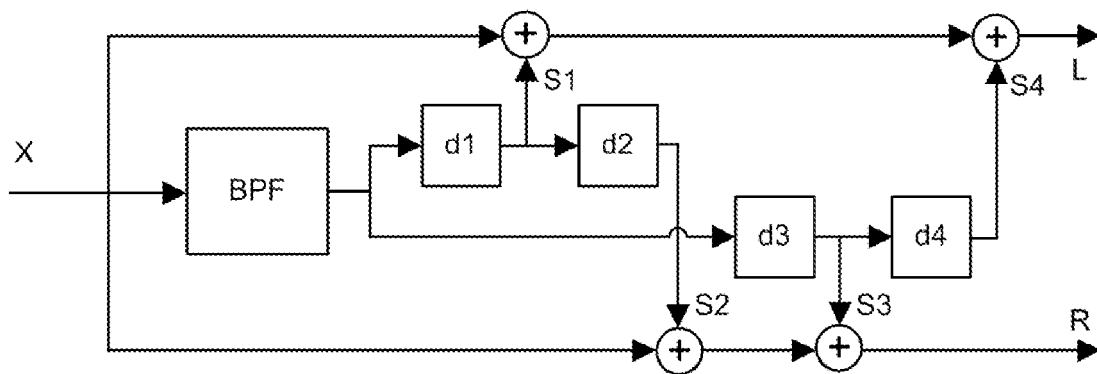


Fig. 1

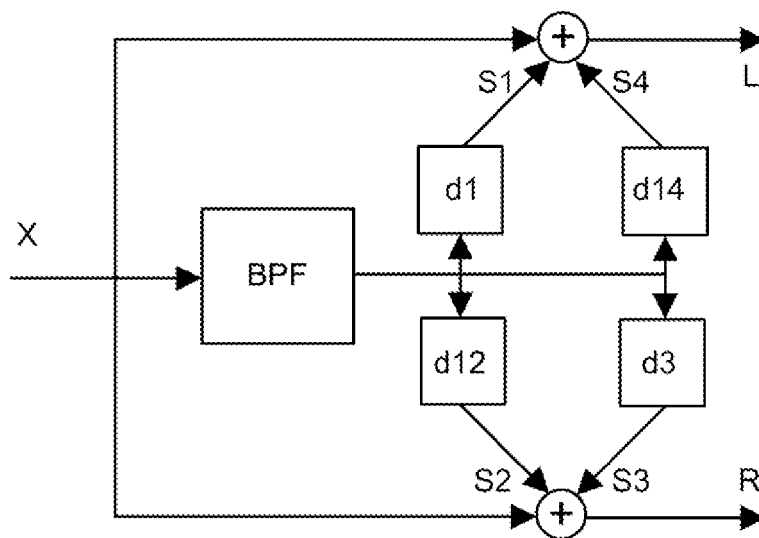


Fig. 2

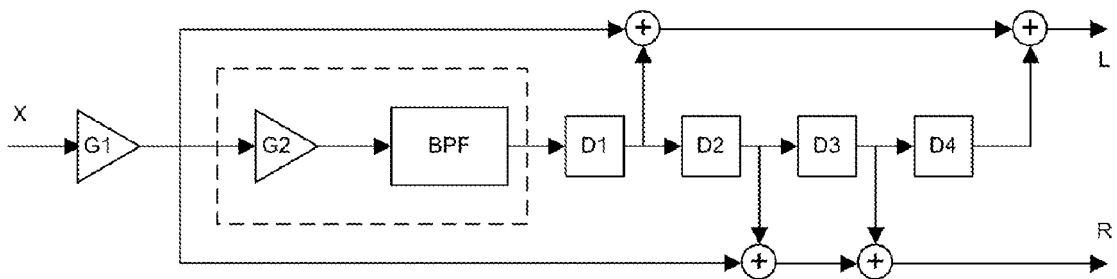


Fig. 3

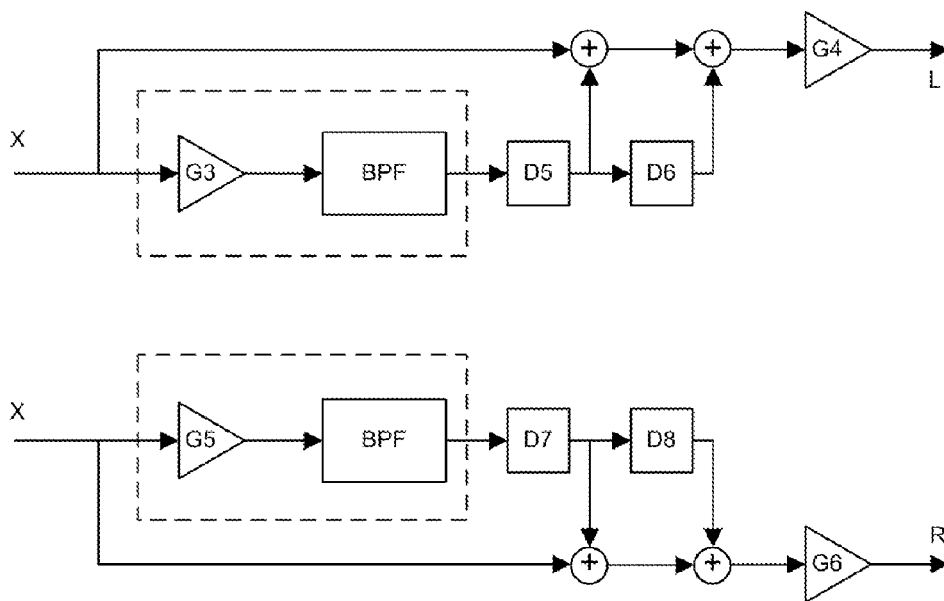


Fig. 4

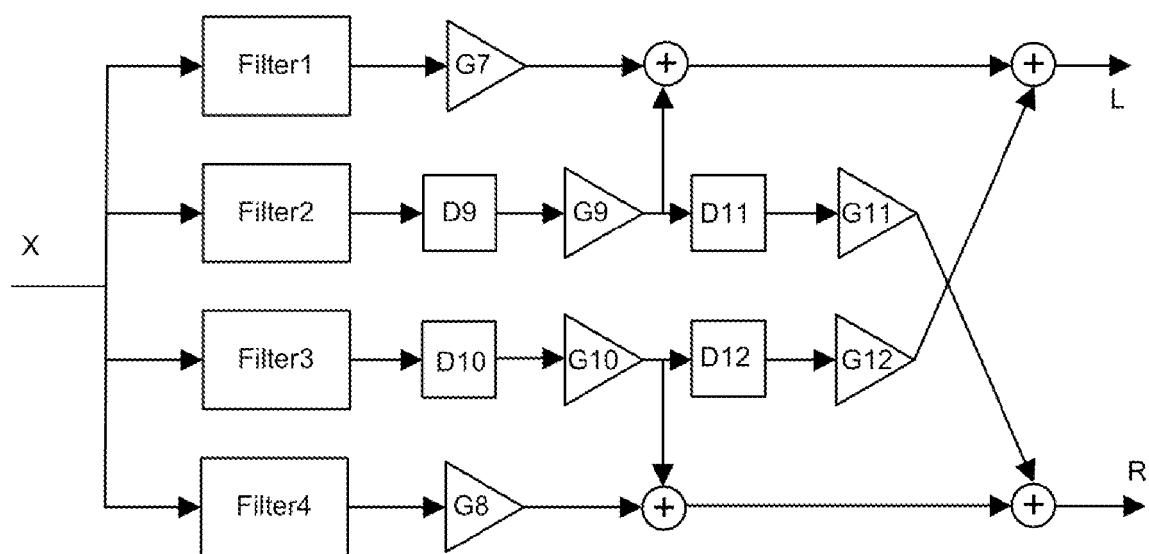


Fig. 5

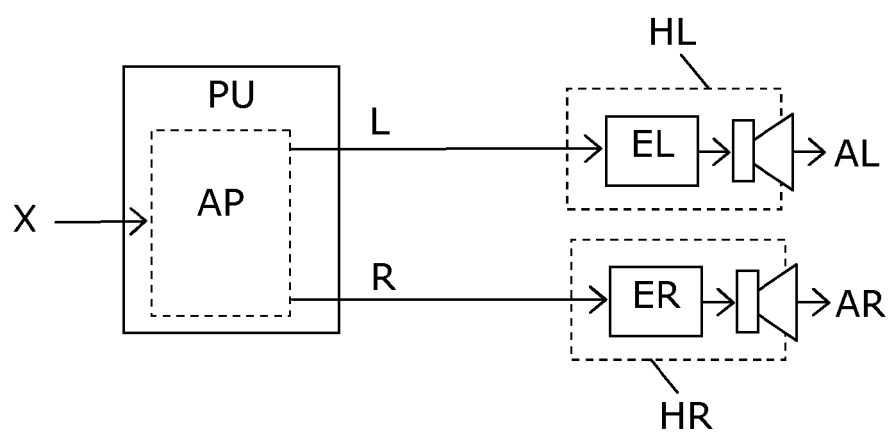


Fig. 6

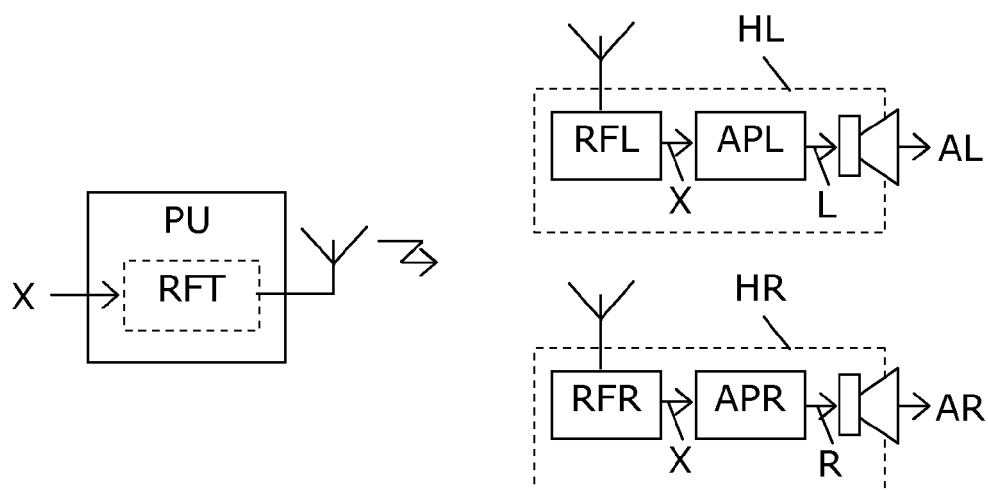


Fig. 7



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EUROPEAN SEARCH REPORT

Application Number
EP 08 10 2617

DOCUMENTS CONSIDERED TO BE RELEVANT			
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
Y	EP 1 365 629 A (SANYO ELECTRIC CO [JP]) 26 November 2003 (2003-11-26) * paragraph [0015] - paragraph [0045]; figures 1,4 *	1-25	INV. H04S5/00
Y	US 2002/015505 A1 (KATZ ROBERT A [US]) 7 February 2002 (2002-02-07) * paragraphs [0012] - [0020], [0050] - [0057], [0110] - [0144] *	1-25	
Y	EP 1 251 717 A (YELLOWKNIFE A V V [AW]) 23 October 2002 (2002-10-23) * paragraphs [0004] - [0010]; figures 1,2 *	1-25	
Y	US 5 173 944 A (BEGAULT DURAND R [US]) 22 December 1992 (1992-12-22) * column 2, line 20 - column 6, line 18; figure 2 *	1-25	
			TECHNICAL FIELDS SEARCHED (IPC)
			H04S
The present search report has been drawn up for all claims			
Place of search Munich		Date of completion of the search 27 August 2008	Examiner Borowski, Michael
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EPO FORM 1503 03.82 (P04C01)

**ANNEX TO THE EUROPEAN SEARCH REPORT
ON EUROPEAN PATENT APPLICATION NO.**

EP 08 10 2617

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The members are as contained in the European Patent Office EDP file on
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27-08-2008

Patent document cited in search report		Publication date	Patent family member(s)	Publication date
EP 1365629	A	26-11-2003	CN 1494812 A	05-05-2004
			WO 02069670 A1	06-09-2002
			JP 3557177 B2	25-08-2004
			JP 2002262398 A	13-09-2002
			US 2005089174 A1	28-04-2005

US 2002015505	A1	07-02-2002	NONE	

EP 1251717	A	23-10-2002	WO 02085067 A1	24-10-2002
			JP 2004527961 T	09-09-2004
			US 2004146166 A1	29-07-2004

US 5173944	A	22-12-1992	EP 0554031 A1	04-08-1993

REFERENCES CITED IN THE DESCRIPTION

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Patent documents cited in the description

- US 6084970 A [0003]

Non-patent literature cited in the description

- **J. Blauert.** Spatial hearing: The psychophysics of human sound localization. MIT Press, 1997 [0004]
- **M. Kahrs ; K. Brandenburg.** Applications of digital signal processing. Kluwer Academic Publishers, 1998 [0005]