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(54) Automatic stereo adaptation

(57) The present invention relates to a playback apparatus for playing back audio signals received by simulcast via different transmission paths. The playback apparatus is particularly adapted to seamless switching between simulcast stereo sound signals being received with different stereo widths. For this purpose, the invention adapts the stereo width of the signals to each other, before switchover. Preferably, the adaptation is performed by attenuating a side signal component of the stereo signal having the larger stereo width.

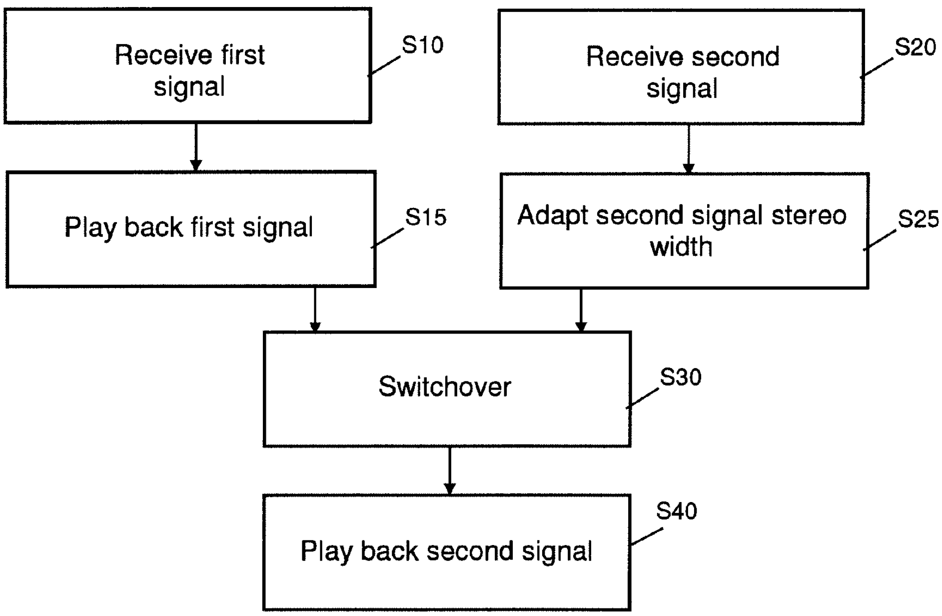

Fig. 1

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Description

[0001] The present invention relates to audio playback. More particularly, the present invention relates to stereo playback by means of an apparatus that is capable of switching over between signals received from different sources.

[0002] Recently, radio apparatuses have been introduced that are capable of receiving signals in multiple ways, broadcast from different sources. In particular, there is the capability of receiving the same signals from different sources. For instance, there can be multiple analogue and/or digital tuners in a single radio receiving apparatus.

[0003] Analogue audio broadcasting systems have been used for a long time in the field of audio broadcasting. Audio broadcasting systems that have been practically employed include AM broadcasting systems in which audio information signals are transmitted in the form of an AM (amplitude modulated) audio information signal and FM audio broadcasting systems in which audio information signals are transmitted in the form of FM (frequency modulated) audio information signals.

[0004] More recently, digital audio broadcasting systems have been introduced, in which audio information signals are transmitted in the form of a digital audio information signal. Thereby, it is generally possible to improve the quality of the audio information as received by a receiver.

[0005] Although it may be expected that digital audio broadcasting systems will become widespread in the future and may replace analogue audio broadcasting systems that have been basically used so far, there will be a certain time period, in which both kinds of systems are used in parallel. In particular, the existing analogue broadcasting systems will remain in service as long as the whole service area thereof is covered by digital broadcasting services. During said time period, it is thus necessary to have analogue and digital audio broadcasting systems operating in parallel at the same time. In particular, one and the same audio service (broadcast program comprising audio signals to be reproduced) will be transmitted by a different (analogue and a digital) broadcasting system (so-called simulcast). While analogue broadcasting systems have been widespread for a long time, and at present they cover a large service area, the service area of the newly emerging digital broadcasting systems has initially been small and will increase only step-wise. Therefore, areas exist which are covered by the service areas of both digital and analogue systems, but in other areas, digital audio broadcasting signals cannot be received, or can be received only with very poor quality, while signals from analogue broadcasting systems can be properly received, thus at the moment enabling a much higher perception quality.

[0006] In the situation as outlined, it is therefore desired that the user can always perceive the audio program with the highest available quality. A radio receiver is therefore capable of switching over between the available systems. A radio receiver normally selects and uses for playback, the signal with best audio quality, at a certain instance in time. Such an issue is of particular importance in the case of mobile audio receivers, for instance, those employed in cars and other vehicles.

[0007] When the said program is available in both different broadcasting systems, a control signal is used for switching the different receivers in order to select the receiver which provides the best reception. In order to achieve a smooth switchover, a cross-fading technique is preferably used therefore.

[0008] It has to be understood that the signals transmitted by different systems such as digital and analogue broadcasting systems, generally have different properties with respect to each other. For instance, signals distributed by digital broadcasting systems have different spectral properties such as bandwidth and volume as compared with the analogue systems. Further, there is a certain time delay between digitally and analogously transmitted signals distributing the same broadcast contents (i.e. the same program of an audio service). In order to achieve a "seamless linking" during switching such that the switching between different sources is not recognized by the listener, radio receiving apparatuses adjust the delays and signal levels between the different sources.

[0009] A further problem arises in the event of reception and playback of stereophonic audio signals. Different stereo broadcast sources, in particular, analogue broadcast sources, are further characterized by different (possibly also reception-dependent) stereo widths of these sources.

[0010] The stereo width is a measure indicating the perceived width of a stereo sound in percent of the speaker distance. When the sound is perceived strictly coming from a single direction, there is a stereo width of zero (mono signal). In case of intensity stereophony (X-Y recording scheme), stereo width increases with an increasing angle between the axes of the recording microphones. In case of time-of-arrival stereophony (A-B recording scheme), stereo width increases with an increasing distance between the recording microphones.

[0011] The stereo width reflects the deviation between the individual signals of the left channel (L) and the right channel (R) constituting a stereo signal. The stereo width can thus be indicated by means of a cross-correlation of left and right channel signals. In the event of fully correlated signals (cross-correlation factor C = 1), the stereo-signal is in fact a mono-signal on two channels and the stereo width is zero. The other extreme is a cross-correlation of -1 (one channel is the inverse of the other), indicating a very large stereo width. The intermediate case in between corresponds to uncorrelated signals (C = 0). Good sounding results are often achieved with values of C between 0.2 and 0.5.

[0012] Stereo signals transmitted (broadcast) with different transmission systems are generally characterized by having different stereo widths. Such a situation is particularly common in the case of the analogue and digital transmissions.

While the stereo width of the digital transmission may be expected to be independent from signal quality, analogue transmission chains often have varying stereo widths.

[0013] It is therefore a drawback of conventional dual apparatuses capable of selecting and switching between stereo signals received from different sources (simulcast) that there is an uncomfortable feeling during the switchover in view of the different stereo widths.

[0014] It is an object of the present invention to provide an improved audio playback method and a respective apparatus that enable seamless switchover between stereo signals for playback that have been received from different sources.

[0015] This is achieved by the features of the independent claims.

[0016] According to a first aspect of the present invention, an audio playback method is provided. The audio playback method comprises the steps of playing back a first stereo signal received from a first source, switching over to playback of a second stereo signal received from a second source, and playing back the second stereo signal after the switchover. Further, the method comprises the step of adapting the stereo width of at least one of the first and the second stereo signals, so as to match the stereo width of the signals to each other, before the switching over step.

[0017] According to a second aspect of the present invention, a stereo playback apparatus is provided. The stereo playback apparatus comprises a first receiving section for receiving a first stereo signal from a first source and a second receiving section for receiving a second stereo signal from a second source. Further, the playback apparatus comprises a playback section for playing back either a first stereo signal received by the first receiving section or a second stereo signal received by the second receiving section. The apparatus further comprises a switching section for switching the played back signal from the first stereo signal received by the first receiving section to the second stereo signal received by the second receiving section. Moreover, the stereo playback apparatus comprises a stereo width adaptation section for adapting the stereo width of at least one of the first and the second stereo signals, so as to match the stereo width of the signals to each other.

[0018] It is the particular approach of the present invention to adapt the stereo width of two audio signals received from different sources to each other. Thereby, a seamless switchover can be achieved between stereo audio signals that provide the same audio service contents, but have different stereo widths, for instance, due to a transmission via different transmission systems. The invention thus enables a comfortable perception in a situation wherein stereo signal switchover in a simulcast environment is necessary, for instance, in a receiver of a vehicle.

[0019] Preferably, one of the stereo signals is an analogue signal and the other stereo signal is a digital signal. In this case, the invention enables a seamless switchover from a digital to a analogue broadcast signal or vice versa, for instance, for a mobile receiver in a case when moving into a service area wherein the digital signal becomes weak or unavailable, or moving back into a digital service area. While the stereo width of the digital transmission is expected to be independent from signal quality, analogue transmission chains often have varying stereo widths. In this case, it is sufficient to realise a stereo adaptation of only one of the signals. The signal with the full stereo width shall be adjusted towards the narrow width of the worse signal.

[0020] The invention is however not limited to the case mentioned above. For instance, a simulcast transmission of two different analogue signals such as AM and FM is a possible field of application of the present invention.

[0021] According to an alternative preferred embodiment, the stereo widths of both signals have been adapted to each other. This can be performed in an iterative procedure.

[0022] Also preferably, the first and second stereo signals respectively transmit one and the same audio service by different broadcast systems.

[0023] Preferably, stereo width adaptation is performed by attenuating a side signal of the at least one of the first and the second stereo signals.

[0024] A side signal corresponds to a channel of a stereo signal that is encoded in accordance with a particular coding method called M/S-coding that is different from the usual stereo encoding employing left (L) and right (R) channels. In accordance with M/S-coding, the stereo channels are separated not into L and R, but into a mid channel (M) and a side channel (S). M/S-stereo coding transforms the left and right channels into a mid channel and a side channel, according to the formulae:

$$M = L + R, \qquad (1a)$$

$$S = L - R \qquad (1b)$$

(possibly additionally scaled by a scaling factor such as 2 or $\sqrt{2}$).

[0025] While the mid channel includes those signal portions that are identically comprised in the L and R channels, the side channel includes those signal portions which are different between L and R. The mid channel thus corresponds to a mono signal for playback, while the side channel indicates the stereo width. A signal without stereo width (mono-signal) thus corresponds to zero signal strength in the side channel S. A back transformation is possible in accordance with the formulae (once more any scaling factors omitted):

$$M + S = 2L \qquad (2a)$$

$$M - S = 2R \qquad (2b)$$

[0026] An example of the use of M/S stereo is FM stereo broadcasting, wherein L + R (M) modulates the carrier wave and L - R (S) modulates a sub carrier. Therefore, in case of analogue FM stereo broadcast, the S-channel is particularly affected under the condition of poor reception quality. Since the side channel indicates the stereo width, the result is a reception-dependent stereo width. The stereo width decreases as the reception quality becomes worse.

[0027] As explained above, a signal encoded in the conventional UR coding can be transformed into M/S-coding, by applying the described calculation scheme. However, a recording scheme (M/S-recording) has also been developed, wherein a stereo signal is directly recording in the form of an M-S signal. In contrast to the usual recording techniques such as intensity stereophony (also known as X-Y technique) and time-of-arrival stereophony (also known as A-B technique), the M/S recording technique (mid-side stereophony) employs a bidirectional microphone facing sideways and another microphone (having an omni-directional or cardioid characteristic), facing the sound source. The mid and side signals are recorded, and the stereo width can be manipulated after the recording has taken place.

[0028] The mid signal represents those parts of the stereo signal which are equal on both channels, and the side signal represents the differences between both channels. If it is desired to widen the stereo width of the signal, it is necessary to increase the relative amplitude of the side signal. Conversely, the stereo width is decreased (the stereo field is narrowed) by decreasing the relative amplitude of the side signal, i.e. the amplitude of the side signal, relative to the mid signal. In geometric terms, the level ratio from S to M approximately corresponds to the angle between the axes of the two directional microphones employed in a conventional X-Y technique (typically in a range between 90° and 135°). The larger the angle, the larger is the stereo width.

[0029] If a signal that has a large stereo width and is thus to be adapted to another signal having a lower stereo width is already received in M/S-coding, stereo width adaptation can be directly performed by attenuating the S component of the received signal.

[0030] In an alternative embodiment, the signal to be adapted is received in UR coding. Therefore, the signal has to be transformed to M/S-coding, before stereo width adaptation.

[0031] Further preferably, the M/S-coded stereo signal is transformed/re-transformed into UR-coding, after the side signal attenuation has been performed. Thereby, playback on a conventional stereo speaker system is enabled.

[0032] Further preferably, the side signal attenuation is controlled based on a comparison of an L-R-similarity measure of a signal, the stereo width of which is to be adapted with an L-R-similarity measure of a reference stereo signal. Therefore, L-R-similarity measures of both signals are determined, and compared with each other. In a preferred embodiment, simply the signal difference per sample is taken as a similarity measure. The smaller the difference is the higher is the degree of similarity. In an alternative preferred embodiment, L-R cross-correlation functions (employing sampling windows of a predetermined length) are calculated for the signal to be adapted and the reference signal, respectively. The larger the correlation is the higher is the degree of similarity. A person skilled in the art is aware of a plurality of other suitable similarity measures such as a summation of absolute or quadratic L-R-signal differences within a predetermined sampling window.

[0033] More preferably, in a case that only the stereo width of one of the received signals is to be adapted, the other one of the received signals the stereo width of which is not to be adapted is used as the reference signal. For instance, for adapting a large stereo width ("full stereo") digital signal to a narrower stereo width analogue signal, by decreasing the stereo width of the digital signal (attenuating the side signal of the digital signal), the analogue signal is used as a reference signal.

[0034] Further preferably, the side signal attenuation is performed step-wisely. In particular, in the case of a rather large difference in stereo widths between the received stereo signals, an adaptation within a single step would be remarkable to the listener and thus cause discomfort. For instance, in a preferred exemplary embodiment, attenuation can be controlled in steps of 1 dB (corresponding to a "negative amplification" of -1dB). However, smaller or larger steps such as 0.5dB or 2dB or any other size of attenuation steps to be adjusted is equally possible. A stepwise adaption depending on the amount of stereo width to be adapted minimizes the discomfort caused to a listener.

[0035] According to a preferred embodiment, the switchover step is performed by cross-fading the first and the second stereo signal. Thereby, a smooth switchover is achieved, providing a high perceiving comfort to a user.

[0036] Further embodiments of the present invention are the subject matter of the dependent claims.

[0037] Additional features and advantages of the present invention will become apparent from the following and more particular description as illustrated in the accompanying drawings, wherein:

Fig. 1 is a flow chart illustrating the main steps of a playback method in accordance with an embodiment of the present invention,

Fig. 2 is a detailed scheme for an automatic stereo adaptation mechanism in accordance with a particular embodiment of the present invention,

Fig. 3 provides an illustration of a stereo music signal encoded in UR-coding, showing a pronounced difference between left and right channels,

Fig. 4 is an illustration of the stereo signal of Fig. 3 after attenuation of the side signal, and

Fig. 5 is a block scheme of an overall system architecture of a stereo playback apparatus according to an embodiment of the present invention.

[0038] Illustrative embodiments of the present invention will now be described with reference to the drawings.

[0039] Fig. 1 is a flow chart providing a general overview of a method according to the present invention. A first and a second stereo sound signal are received at steps S10 and S20, respectively. Signal reception at steps S10 and S20 is performed in parallel, at the same time. Typically, the first and second received signals provide the same contents in multiple ways (via multiple transmission systems), such as analogue and digital, or via different analogue transmission systems such as AM and FM, or different digital transmission systems such as DRM (Digital Radio Mondiale) and DAB (Digital Audio Broadcast), or transmitted via satellite and terrestrially.

[0040] Although a radio receiver in accordance with the present invention is generally capable of receiving two audio signals in parallel, only one of these signals is selected for playback, at a certain instance of time. In accordance with the flow chart of Fig. 1, at an initial instance of time the first signal is selected and played back (step S15). Generally, a radio receiver selects the particular one out of the two signals received which enables the highest quality of perception at the moment of selection. In the case of parallel digital and analogue signal reception, it may generally be assumed that better quality is provided by the digital signal. However, this is true only as long as the receiver is situated within a high quality service area of the digital broadcast. In case the receiver is moved (for instance, by driving a vehicle) outside said service area, an analogue signal, which remains to have a larger service area for a certain period of time, may be able to provide a better quality. In this case, switchover to the analogue signal is desired. Vice versa, when the vehicle drives back into the high quality service area of the digital signal, it may be expected that a better service quality is provided by the digital signal, so that a switchover back to the digital signal is desired. The receiver detects the current reception quality of the received signals, and decides which signal to select for playback at a certain instance of time.

[0041] For good quality audio receivers, it is however further desired that the switching between different sources shall not be recognized by the listener. Therefore, delays and differences in signal levels between the received sound signals are adjusted (not shown in the flow chart of Fig. 1). In accordance with the particular approach of the present invention, in the case of stereo sound signals, stereo widths of the signals are further adapted to each other before performing the switchover. In many cases, it is sufficient to adapt only the stereo width of one of the signals, by reduction or enhancement. Usually, the signal having the larger stereo width is adapted to the signal having the lower stereo width, by stereo width reduction. In the particular case illustrated in Fig. 1, the stereo width of the second signal is adapted at step S25.

[0042] In practical implementation, such a situation occurs, for instance, if playback is switched from an analogue signal to a digital signal. The digital signal generally and stably has a large stereo width. By adaptation of the stereo width at step S25, switchover is performed in step S30 with signals, the stereo widths of which coincide to a high degree of accuracy, at a lower level. After switchover, the second stereo sound signal is played back at step S40. To benefit from the larger stereo width of the second received signal, it is possible to increase the stereo width for playback at a later stage, after the switchover has been performed.

[0043] As indicated above, the present invention is however not limited to the particular situation shown in Fig. 1. To the contrary, the invention also covers a situation wherein the stereo width of the signal that is initially played back ("first signal") has to be adapted before switchover. For instance, such a situation is common in the case of switchover from a digital to an analogue broadcast signal. The stereo width of the first (digital) signal (being generally large) is decreased before switchover, i.e. while the first (digital) signal is still selected for playback. Switchover is performed when the stereo width of the analogue (second) signal has been reached by the digital signal. In the same manner as in the illustrated embodiment, switchover is performed between signals having coinciding (low) stereo width, so that no difference in stereo width is perceivable by the listener during switchover.

[0044] Further, the present invention also covers a situation wherein the stereo width of both received signals is adapted before switchover, i.e. the (low) stereo width of one of the signals is increased and the (high) stereo width of the other signal is decreased, to an intermediate stereo width.

[0045] Fig. 2 is an illustration of an automatic stereo adaptation mechanism according to an embodiment of the present invention. In the mechanism described with reference to Fig. 2, the stereo width of only one received signal is adjusted. Such a mechanism is sufficient in most cases.

[0046] As illustrated in the upper portion of Fig. 2, UR-coded stereo sound signals are provided. On the left hand side of the drawing, signal 100 to be adjusted is illustrated. The right hand side of the drawing illustrates the other received signal, the stereo width of which remains unchanged. The described mechanism however employs said signal as a reference signal 200 for adjusting the stereo width of signal 100. The drawing of Fig. 2 shows the complete signal chain from two stereo inputs to a cross-faded output. It is assumed that signal level adaptation and delay compensation have been performed before.

[0047] The stereo adaptation mechanism of Fig. 2 starts with receiving a stereo sound signal 100 to be adjusted and another stereo sound signal 200 that is used as a reference signal for stereo width adjustment. The received stereo signal 100 is re-encoded into M/S-coding by performing adding 102a and subtracting 102b operations of the left L and right R channels of the received signal 100. As already briefly described above, there is a possibility that the signal to be adjusted is already received in M/S-coding. In this case, the first operation step of the mechanism (102) can be left out.

[0048] After M/S-re-encoding, side signal S is attenuated. As explained above, this will reduce the stereo width of the signal. Attenuation 104 is controlled by stereo width stepping logic 310, as will be detailed below. It has to be noted that attenuation by attenuator 104 can be bypassed at an initial stage of the mechanism, as the stereo width stepping logic 310 has not yet been provided with control information.

[0049] After the adjustment, the signal is again transferred to UR-coding by adding 106a and subtracting 106b signals M and S. This is followed by an optional step of a uniform attenuation of both re-established L and R signals by attenuators 108a and 108b, respectively. The illustrated attenuation by 6dB corresponds to a compensation of the factor of two in the above indicated formulae (2a) and (2b) for back transformation from M/S to UR.

[0050] Subsequently, UR similarity of the processed signal is calculated by similarity calculator 110 (in the illustrated case: simply a per sample difference, which actually corresponds to a signal level of the side signal is calculated). A respective UR similarity of the reference signal 200 is calculated by similarity calculator 210. The adjusted signal and the reference signal are then compared in comparator 300. Comparator 300 calculates the difference of both UR similarity measures determined by calculators 110 and 210, respectively.

[0051] The thus determined similarity difference 315 is input to stereo width stepping logic 310. Stereo width stepping logic 310 averages the difference 315, in order to provide a stable control basis.

[0052] As indicated above, the employed similarity measure is however not limited to the simple case of a per sample difference (side signal strength) illustrated in Fig. 2. For instance, a L-R cross-correlation function can be employed which has to be calculated separately for both signals (by replacing subtractors 110 and 210 with correlators respectively). In that case, similarity difference 315 would be represented by a correlation difference. As described in the introductory portion of the specification, the UR correlation difference is generally a measure indicating a difference in stereo width between the signals.

[0053] When similarity difference 315 is zero, this means that no difference in stereo width exists between the two compared signals. Consequently, no further adaptation would be necessary. On the other hand, similarity difference 315 indicates an amount by which side signal S has to be attenuated, in order to adapt the stereo width of signal 100 to that of reference signal 200. In order to enable a smooth stereo width adaptation that is not perceivable by the user, the adaptation is preferably performed in a stepwise manner.

[0054] In the simplest case, stereo width stepping logic 310 instructs attenuator 104 to attenuate side signal S by an adjustable coefficient, which is stepwisely increased as long as a similarity difference 315 is still detected. For instance, the attenuation coefficient of attenuator 104 can be adjusted in steps of 1dB or 0.5dB, 2dB or any other amount within or even outside said range. An attenuation by 1dB corresponds to a "negative amplification" by an amplification coefficient corresponding to -1dB. An attenuation by 1dB means that the resulting signal strength is approximately 11% weaker than the signal strength before attenuation (or "amplified" by a factor of 0.89). After the similarity difference 315 of zero has been determined by comparator 300, the attenuation coefficient of attenuator 104 will be kept constant (as long as no similarity difference occurs once more, due to variations in received signal stereo widths).

[0055] Thus, adjustment of the stereo width is done, in principle, until the difference becomes zero. In practice, the accuracy that can be achieved is given by the size of the steps in which the attenuation coefficients can be controlled. The attenuation is increased until the similarity difference 315 reaches a minimum. In particular, when the difference in the applied similarity measure starts to increase after increasing the attenuation in a current step (i.e. the minimum in the difference 315 has been passed through), the amount of attenuation has to be again decreased, thus leading to a smaller degree of attenuation. It is sufficient to achieve a difference in stereo widths that is lower than a predetermined upper boundary after the stereo adaptation has been performed, such that a cross-fading between the signals can be done without audible differences in the stereo width.

[0056] In accordance with another embodiment, stepwise reduction of the signal strength can be achieved by controlling attenuation in such a way that a predetermined percentage such as 20% or 50% of the similarity difference 315 is reduced in one step. The percentage values are however not limited to those indicated by way of example, but any percentage such as in a range between 10% and 50%, or even outside this range is possible.

[0057] Switchover is performed by cross-fading the signals in a conventional manner. Thereby, during switchover both received signals are overlaid by adders 330a (L channel) and 330b (R channel), respectively, with different weights determined by attenuators 112a, 112b, 212a and 212b. The weights given to the signals are therefore varied during the cross-fading switchover procedure, under the control of cross-fading logic 320. The cross-fading is preferably controlled so as to keep a constant output level. At the beginning (corresponding to S15 of Fig. 1), the first of the signals (either processed signal 100 or reference signal 200) is output without further attenuation (weight 1), while the other one of the signals is attenuated to zero. At the end of the cross-fading switchover (corresponding to step S40 of Fig. 1), the situation is reversed. The initially selected signal is further attenuated to zero, while the signal that has initially been attenuated to zero is given weight 1 by being output without further attenuation. In the meantime, both weights are varied continuously or in small steps from one to zero or from zero to one, respectively, while keeping an output level constant. Thereby, a smooth switchover is achieved. As a result, a switchover of output signal 400 is realised. It has to be emphasized that the cross-fading switchover illustrated in Fig. 2 does not depend on whether signal 100 that is adjusted is the signal that is initially selected for playback ("first signal") or the signal selected for playback after switchover ("second signal").

[0058] As further indicated above, the present invention is not limited to the case described with reference to Fig. 2, wherein only a single one of the received stereo sound signals is to be adjusted. To the contrary, in a similar scheme as the one described, also a stereo width adjustment of two signals towards each other may be achieved. Specifically, a similarity difference between two stereo signals can be minimized, by attenuating the side signal of one of the signals and at the same time increasing the side signal of the other one of the signals. Therefore, both received signals have to be intermediately encoded according to the M/S-coding scheme. Further alternatively, for instance also an adjustment by only increasing the side signal of one of the signals is possible.

[0059] The difference in stereo widths that can be reached by attenuating the side signal is illustrated with reference to Figs. 3 and 4. Both Figs. 3 and 4 show a stereo signal of a piece of music over the time axis, in UR-coding. The L-channel signal is shown in the upper portion of Figs. 3 and 4. The signal illustrated in the lower portion of Figs 3 and 4 is the signal of the R-channel. In Fig. 3, the difference between left and right is clearly visible. The result of an attenuation of the side signal, after re-coding to UR, can be seen in Fig. 4. Fig. 4 shows the reduced difference between left and right.

[0060] Fig. 5 is a block scheme of an exemplary stereo playback apparatus according to the present invention. The playback apparatus comprises a first antenna 10, a second antenna 20, a first signal reception portion 12 and a second signal reception portion 22, a stereo width adaptation portion 14 for the signal received by the first signal reception portion 12, a stereo width adaptation portion 24 for the signal received by the second signal reception portion 22, a selection switch 30 a playback portion 40 and a control portion 50. The playback apparatus may further include circuitry for delay compensation between signals received by the first and the second reception portions respectively, and for signal level adaptation. These components are however not essential for the present invention, and therefore have not been illustrated in Fig. 5.

[0061] Control portion 50 controls the operation of switching portion 30 for selecting one of the signals received by the first and the second signal reception portions 12 and 22 respectively. Preferably, one and the same audio service is received via two different transmission paths (simulcast). As described in detail above, a selection of a signal for playback can thereby be achieved, which provides the best audio perception quality available at a particular instance of time. Control section 50 evaluates the received signals, and controls switching section 30 so as to always select and play back the signal enabling the best perception quality to a listener.

[0062] Switching section 30 can be realised by means of a cross-fader, enabling a smooth switchover between the two signals.

[0063] Control section 50 further controls stereo width adaptation portions 14 and 24 so as to minimize the perception of the switchover by the user, by adapting the respective stereo widths. In particular, control portion 50 preferably includes a left/right similarity calculation portion such as a correlation calculation portion for both received signals, and a logic for calculating a difference thereof. Based thereon, it is decided whether there is a difference in stereo widths of the two received signals, and an adaptation of the stereo width of at least one of the signals is necessary. If control section 50 decides that no stereo width adaptation is necessary for a particular one of the received signals, a respective stereo width adaptation portion 14 or 24 can be controlled to be bypassed by the respective signal.

[0064] After stereo width adaptation, there is no noticeable difference between the stereo width of the signals, and switching between the two signals can be performed inaudibly at an time. The present invention as defined by the appended claims is not limited to those particular embodiments that have been described in detail above. A person skilled in the art is aware of further modifications of the described embodiments.

[0065] In summary, the present invention relates to a playback apparatus for playing back audio signals received by simulcast via different transmission paths. The playback apparatus is particularly adapted to seamless switching between simulcast stereo sound signals being received with different stereo widths. For this purpose, the invention adapts the stereo width of the signals to each other, before switchover. Preferably, the adaptation is performed by attenuating a side signal component of the stereo signal having the larger stereo width.

Claims

1. An audio playback method, comprising the steps of:

    playing back (S15) a first stereo signal (100, 200) received from a first source,
    switching over (S30) to playback of a second stereo signal (100, 200) received from a second source, and
    playing back (S40) said second stereo signal (100, 200) after the switchover,

    characterized by the step of
    adapting (S25) the stereo width of at least one (100) of the first and the second stereo signals (100, 200) so as to match the stereo widths of said signals to each other, before said switching over step (S30).

2. A method according to claim 1, wherein said adapting step (S25) comprising the step of attenuating (104) a side signal (S) of said at least one (100) of the first and the second stereo signals.

3. A method according to claim 2, wherein said side signal attenuating step (104) is performed stepwisely.

4. A method according to claim 2 or 3, wherein said signal (100) the stereo width of which is to be adapted is UR-coded, the method further comprising the step of transforming (102a, 102b) said UR-coded stereo signal (100) into an M/S-coded stereo signal.

5. A method according to any of claims 2 to 4, further comprising the step of transforming (106a, 106b) said stereo signal into UR-coding, after said side signal attenuating step (104).

6. A method according to any of claims 1 to 5, further comprising the steps of
    determining (110) an L-R-similarity measure of said signal (100) the stereo width of which is to be adapted,
    determining (210) an L-R-similarity measure of a reference stereo signal (200), comparing (300) said determined L-R-similarity measures with each other, and controlling said adapting step (S25) based on the comparison result (315).

7. A method according to claim 6, wherein said reference signal (200) is one of said first received and said second received stereo signal, other than the one (100) the stereo width of which is adapted.

8. A method according to any of claims 1 to 7, wherein said switchover step (S30) including the step of cross-fading said first stereo signal and said second stereo signal (100, 200).

9. A stereo playback apparatus, comprising
    a first receiving section (10, 12) for receiving a first stereo signal (100, 200) from a first source,
    a second receiving section (20, 22) for receiving a second stereo signal (100, 200) from a second source,
    a playback section (40) for playing back either a first stereo signal received by the first receiving section (10, 12) or a second stereo signal received by the second receiving section (20, 22), and
    a switching section (30) for switching the played back signal from the first stereo signal received by said first receiving section (10, 12) to the second stereo signal received by said second receiving section (20, 22),
    characterized by
    a stereo width adaptation section (14, 24) for adapting the stereo width of at least one (100) of the first and the second stereo signals (100, 200) so as to match the stereo widths of said signals to each other.

10. A stereo playback apparatus according to claim 9, wherein said stereo width adaptation section (14, 24) comprising an attenuator (104) for attenuating a side signal (S) of said at least one (100) of the first and the second stereo signals.

11. A stereo playback apparatus according to claim 10, wherein said signal (100) the stereo width of which is to be adapted is UR-coded,
the apparatus further comprising transforming means (102a, 102b) for transforming said UR-coded stereo signal (100) into an M/S-coded stereo signal.

12. A stereo playback apparatus according to claim 10 or 11, further comprising transforming means (106a, 106b) for transforming the stereo signal after said side signal attenuation into UR-coding.

13. A stereo playback apparatus according to any of claims 9 to 12, further comprising
a first similarity calculator (110) for determining an L-R-similarity measure of said signal (100) the stereo width of which is to be adapted,
a second similarity calculator (210) for determining an L-R-similarity of a reference stereo signal (200),
a comparator (300) for comparing said determined L-R-similarity measures with each other, and
a control section (310) for controlling said stereo width adaptation section (14, 24) based on the comparison result (315) of said comparator (300).

14. A stereo playback apparatus according to claim 13, wherein said reference signal (200) is one of said first received and said second received stereo signal, other than the one (100) the stereo width of which is adapted.

15. A stereo playback apparatus according to any of claims 9 to 14, wherein said switching section (30) being adapted to cross-fade said first stereo signal and said second stereo signal (100, 200).

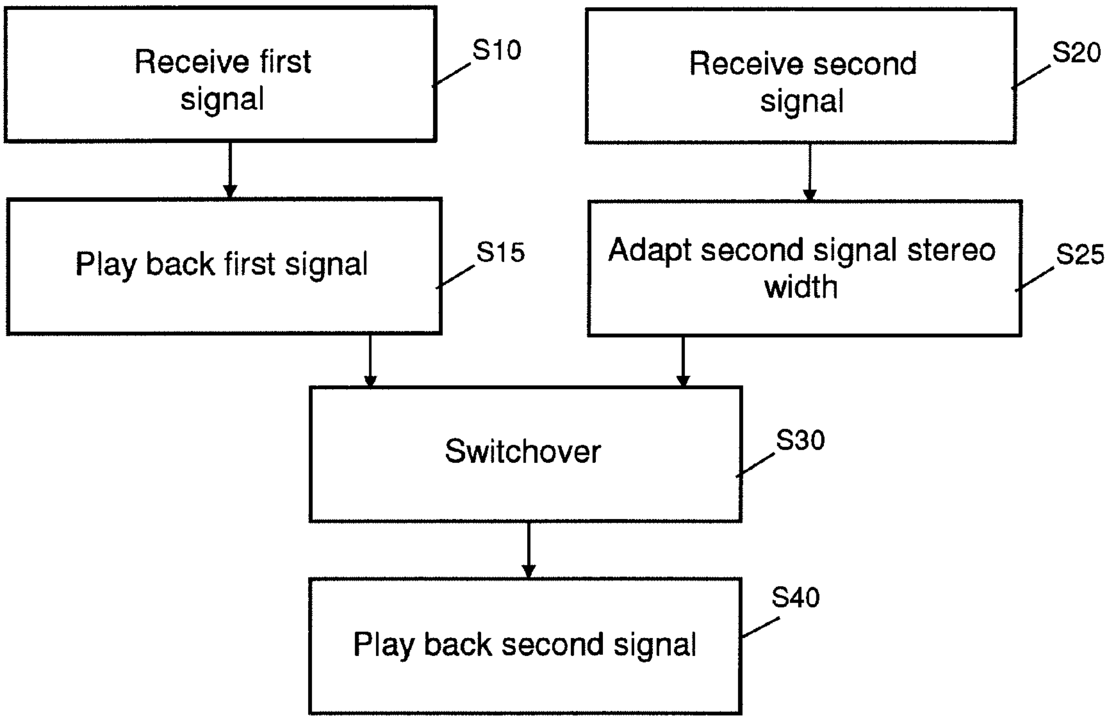

Receive first signal
S10
Receive second signal
S20
Play back first signal
S15
Adapt second signal stereo width
S25
Switchover
S30
Play back second signal
S40

Fig. 1

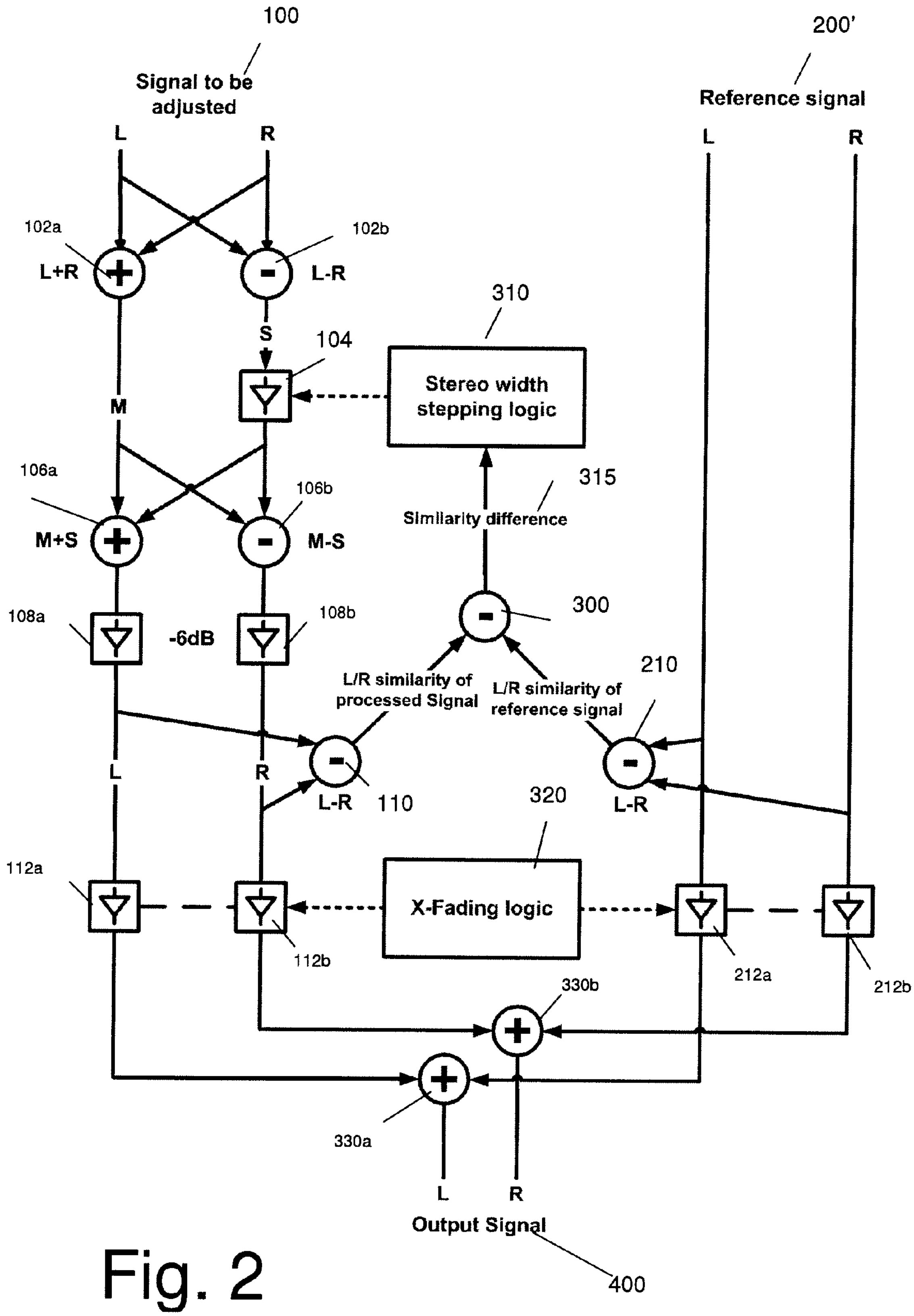

100
Signal to be adjusted
L
R
200'
Reference signal
L
R
102a
L+R
102b
L-R
S
104
310
Stereo width stepping logic
M
106a
106b
M+S
M-S
315
Similarity difference
108a
-6dB
108b
300
L/R similarity of processed Signal
L/R similarity of reference signal
210
L
R
L-R
110
320
L-R
112a
X-Fading logic
112b
212a
212b
330b
330a
L
R
Output Signal
400

Fig. 2

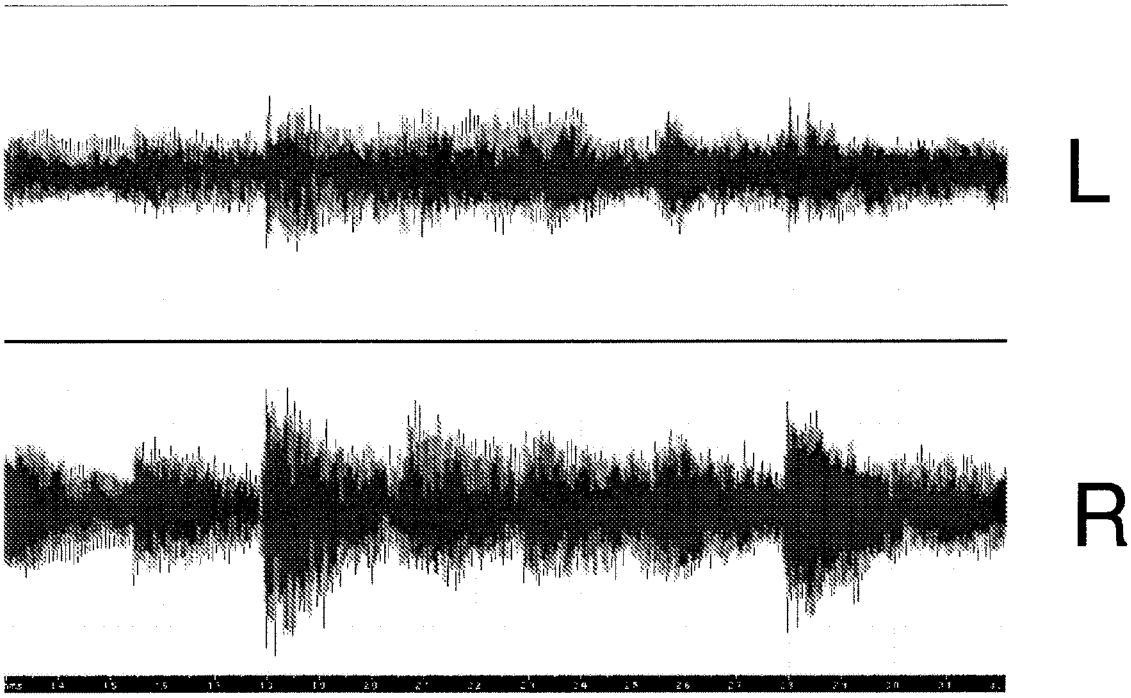


Fig. 3

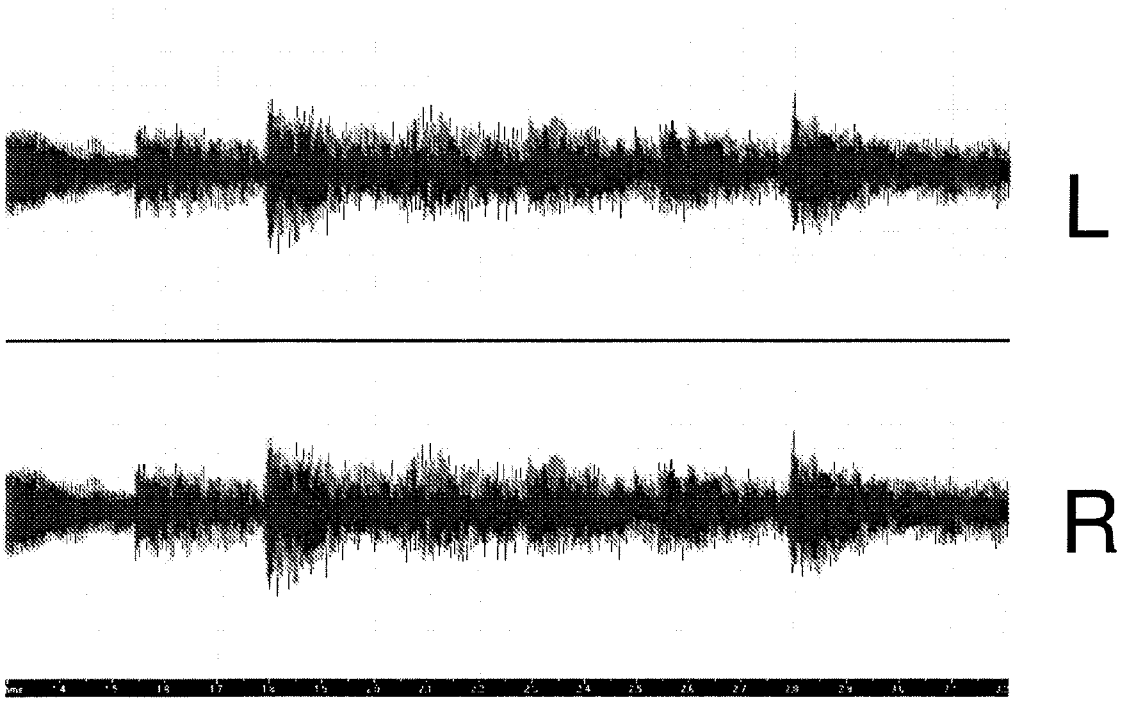
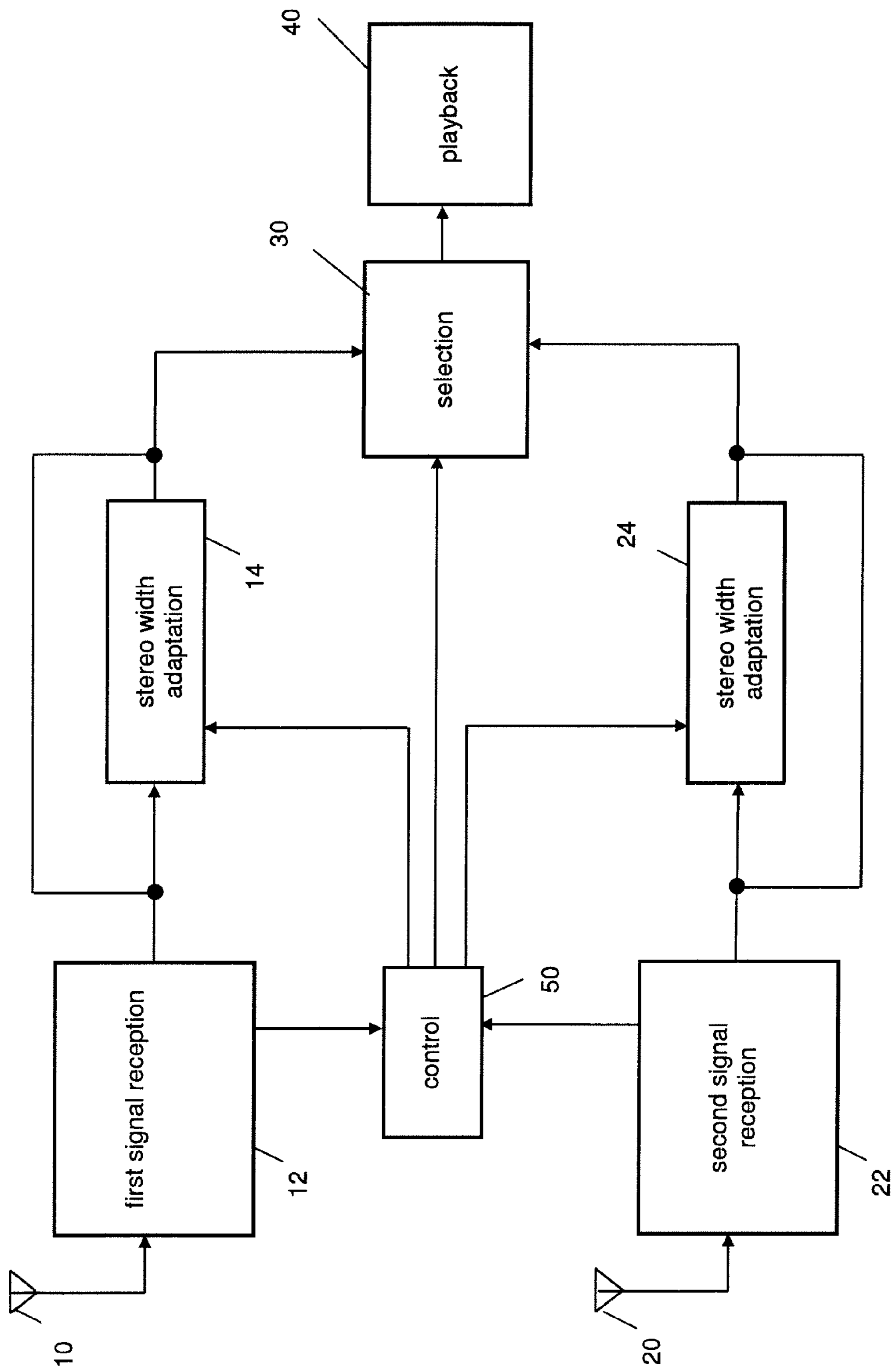


Fig. 4



10
first signal reception
12
20
second signal reception
22
control
50
stereo width adaptation
14
stereo width adaptation
24
selection
30
playback
40

Fig. 5

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EUROPEAN SEARCH REPORT

Application Number
EP 10 17 9424

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A	* page 2, line 11 - page 3, line 14 * * page 5, line 12 - page 6, line 2 * * page 10, line 15 - page 11, line 25 * * page 12, line 16 - page 13, line 11 * -----	2,7,10, 14	
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A	US 2003/129941 A1 (KAWAMATA YUKIHIRO [JP] ET AL) 10 July 2003 (2003-07-10) * abstract * * paragraph [0006] - paragraph [0008] * * paragraph [0049] - paragraph [0071] * * paragraph [0115] - paragraph [0139] * -----	1-15	

1 The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
The Hague	3 December 2010	Zanti, Patrizio

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03-12-2010

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