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(54) Optimized delay detection of simulcast broadcast signals

(57) The present invention relates to a particular efficient scheme for determining an initial delay of a second received audio signal with respect to a first received audio signal, specifically suitable for a simulcast environment. In order to receive a seamless and inaudible switchover in a simulcast environment, the initial delay time must be exactly determined and compensated by delaying the first signal at the receiving apparatus by a respective amount. Calculation power requirements are particularly high, if the sound signals reproduced from first and second received audio signals have different sampling frequencies. The present invention adapts the sampling frequencies by downsampling both signals for delay time determination and thus considerably reduces calculation effort.

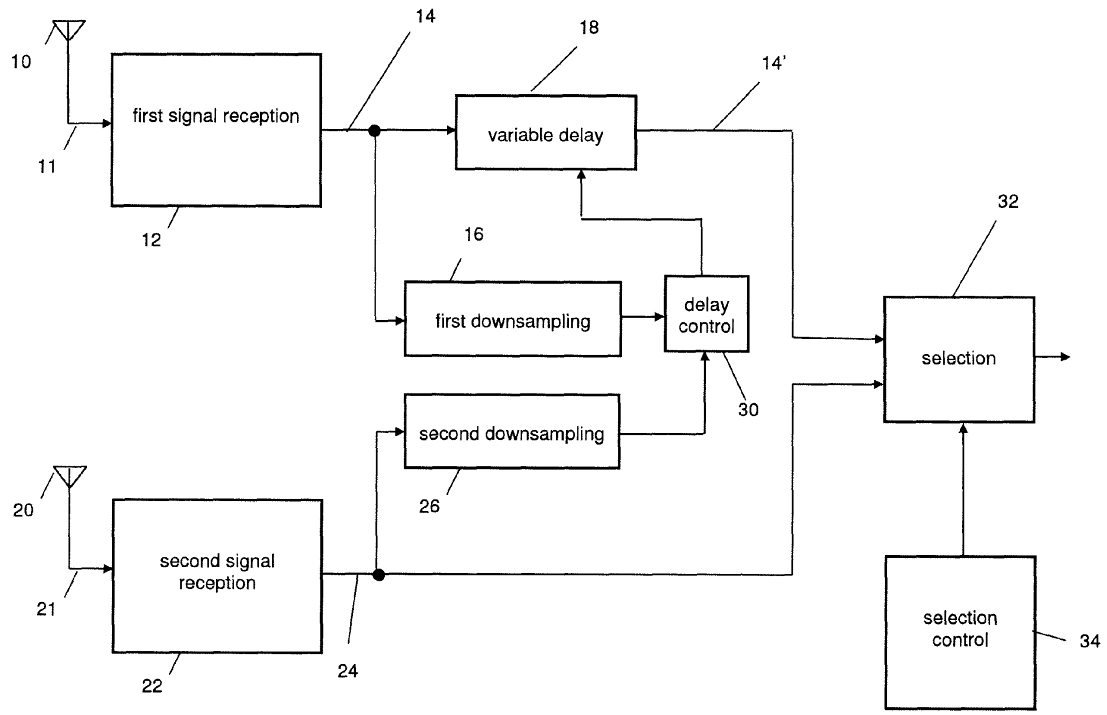

Fig. 1

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Description

[0001] The present invention relates to an apparatus for receiving broadcast signals. More particularly, the present invention relates to a broadcast signal receiving apparatus which is operative to receive broadcast signals that are transmitted via two different broadcast systems.

[0002] 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 audio 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 signal.

[0003] 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. For instance, in European countries, a digital audio broadcasting system called DAB (Digital Audio Broadcasting) is put into service. Another example for a digital audio broadcasting system is DRM (Digital Radio Mondiale).

[0004] Although it may be expected that digital audio broadcasting systems will become widespread in the future, and mainly 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 via different (analogue and digital) broadcast systems. While analogue broadcast systems have been widespread for a long time and therefore at the present time cover a large service area, the service area of the newly emerging digital broadcasting systems has initially been small and will increase only step-wisely. Therefore, areas exist which are covered by the service areas of both digital and analogue systems, but in other areas digital audio broadcast signals cannot be received, or can be received only in very poor quality, while signals from analogue broadcasting systems can be properly received, thus at the moment enabling a much higher perception quality.

[0005] In the situation as outlined, it is therefore desired that a user can always perceive the audio program with the highest available quality. In case of supply bottlenecks, it is therefore necessary to switch over between the available systems, in order to always select the system for reception, which delivers the best audio quality, at a certain instance of time. Such an issue is of particular importance in the case of mobile audio receivers, for instance those employed in cars and other vehicles.

[0006] In order to always achieve the best possible reception quality, audio receivers have been developed, which include both analogue and digital receiving sections. When the same program is available on both of the different broadcasting systems, a control signal is used for switching the different receivers in order to select the receiver which provides the best reception.

[0007] 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 via digital broadcasting systems have different spectral properties such as bandwidth and volume as compared with the analogue systems. A further particularly important difference between digitally and analogously transmitted signals is a certain time delay of the digital signals with respect to the analogue signals distributing the same broadcast contents (i.e. the same program of an audio service).

[0008] The reason for said time delay resides in the particulars of the digital broadcast transmission chain. The digital coding results in a noticeable delay of a digital broadcast audio signal compared to the corresponding analogue audio signal. One important source of coding delays in the digital broadcast system is, for instance, time interleaving of audio information data for the purpose of minimizing deterioration. A typical delay of a digital audio broadcast signal with respect to an analogue broadcast signal of the same service is in the order of several hundreds of milliseconds (ms), for example, 400 ms, up to the order of one second (s), for instance 1.4s.

[0009] In the case, when a switchover from a first broadcasting system to a second broadcasting system becomes necessary, for instance, when the digital broadcast signal becomes weak due to a movement of a vehicle out of a service area of the digital broadcasting station, it is therefore desired to perform the switchover from the digital broadcasting system to the analogue broadcasting system for reproduction in such a manner that the user takes as little notice of the switchover as possible. In order to switchover between two broadcasting systems having the same audio contents, it is therefore necessary to analyse the main differences between the audio signals, and to perform a respective adaptation for the switchover. In particular, a compensation of the time delay is necessary.

[0010] Generally, the delay between the analogue and the digital transmission varies from broadcasting station to broadcasting station. It is therefore not possible to perform the delay time compensation based on the assumption of a predetermined delay value. On the contrary, the delay has to be dynamically determined during operation, in the audio receiver.

[0011] A known apparatus for receiving audio broadcast signals is capable of receiving both digital audio broadcast signals and analogue audio broadcast signals by means of respective reception portions, to obtain a first reproduced sound signal, from the digital broadcast signal and a second reproduced sound signal from the analogue broadcast signal, respectively. Further, the apparatus comprises a variable delay portion for delaying the first reproduced sound signal obtained from the analogue audio broadcast signal by a variable delay time. The variable time delay applied by the variable delay portion is controlled so as to reduce a difference in delay time between the first reproduced time signal after being delayed by the variable delay portion and the second reproduced sound signal. Such a conventional apparatus for receiving both analogue and digital broadcasting systems is known from European patent EP 0 863 632 B1.

[0012] A similar conventional apparatus is known from document EP 1 227 608 A2, wherein two signal matching control portions are employed for controlling the variable delay between the two audio signals and the results of the control portions are combined. The apparatus additionally also takes into account an adaptation of further differences between the sound signals reproduced from digital and analogue sources, such as volume.

[0013] For further processing, also the reproduced sound signals from the analogously received broadcast signals are generally digitalized so that both reproduced sound signals are available in digitalized form, however, generally having different sampling frequencies.

[0014] It is a drawback of prior art receivers for signals from different broadcasting systems that the control processing for delay compensation is complicated, in particular, for considering different sampling frequencies of the digitalized analogue broadcast signal and digital broadcast signal. Respective control portions are therefore complicated and rather costly. Although it is generally possible to implement the control portions in the form of a digital signal processor, however, a significant calculation time would be required.

[0015] A problem in efficiently calculating a delay between broadcast signals received via different broadcast systems does not only occur in case of parallel reception of certain audio contents via analogue and digital transmissions such as FM and DAB or AM and DRM. Similar problems occur in any broadcast environment, wherein one audio service is transmitted via different broadcast systems (a so-called "simulcast transmission"), and wherein an inaudible switching between the two signals shall be performed. Further examples for such a situation include, but are not limited to: DRM and DAB simulcast transmissions, analogue AM or FM and IBOC (In-Band-On-Channel) simulcast transmissions, and satellite and terrestrial audio broadcast systems in general.

[0016] The present invention aims to provide an improved audio receiver and a respective receiving method enabling a more efficient control of the delay between two audio signals of different sampling rates originating from different broadcast sources.

[0017] This is achieved by the features of the independent claims.

[0018] According to the first aspect of the present invention, an audio receiver is provided. The audio receiver comprises a first signal receiving portion for receiving a first audio broadcast signal to obtain a first reproduced sound signal, and a second signal receiving portion for receiving a second audio broadcast signal to obtain a second reproduced sound signal. The audio receiver further comprises a variable delay portion for delaying the first reproduced sound signal by a variable delay time. Moreover, the receiver comprises a delay control portion for controlling the variable delay portion. The audio receiver further comprises a first downsampling portion for downsampling the first reproduced sound signal and a second downsampling portion for downsampling the second reproduced sound signal. The delay control portion controls the variable delay portion based on the first and second reproduced sound signals downsampled by the first and second downsampling portions.

[0019] According to a second aspect of the present invention, a method of receiving audio broadcast signals is provided. The method comprises the steps of receiving a first audio broadcast signal, obtaining a first reproduced sound signal from said received first audio broadcast signal, receiving a second audio broadcast signal and obtaining a second reproduced sound signal from the received second audio broadcast signal. Further, the method comprises the step of delaying the first reproduced sound signal by a variable delay time. The method further includes the steps of downsampling the first reproduced sound signal and downsampling the second reproduced sound signal, and controlling the variable delay time based on the downsampled first and second reproduced sound signals.

[0020] It is the particular approach of the present invention to control a variable delay portion for delaying one out of two sound signals reproduced from two audio broadcast signals received in an audio receiver, based on downsampled versions of the respective sound signals. The invention thereby enables to precisely determine an initial delay between the two signals with low calculation effort. Thereby, the initial delay can be efficiently compensated for, in particular, for sound signals originating from simulcast transmissions and having different sampling frequencies.

[0021] Preferably, the first audio broadcast signal is an analogue signal and the second audio broadcast signal is a digital signal. In compliance with this preferred embodiment, the invention enables a seamless switchover from a digital to an analogue broadcast signal, for instance, for a mobile receiver in case when moving into a service area wherein the digital signal becomes weak or unavailable. Due to coding delays in the digital broadcast transmission chain, the digital broadcast audio signal has a noticeable delay compared to an analogue audio signal of the same service, which shall be compensated for to achieve seamless switchover.

[0022] The invention is, however, not limited to the case mentioned above. For instance, in a simulcast transmission, the analogue signal can be given an initial delay when sent out from the broadcast station. In that case, the first audio broadcast signal, according to the invention, can be a digital signal, and the second signal can be an analogue signal. Further examples of signals to which the present invention is applicable include, but are not limited to cases, wherein either one of the first and second audio broadcast signals are DRM and DAB or analogue AM and DRM, or analogue AM/FM and IBOC.

[0023] Also preferably, one of the audio broadcast signals is transmitted via satellite and the other is transmitted via terrestrial transmission. In particular, the signal transmitted via satellite can generally be assumed to be received with an initial delay with respect to a corresponding terrestrially transmitted signal, due to the considerably larger travelling distance of the signal. In that case, the signal transmitted terrestrially corresponds to the first signal and shall be delayed at the receiver, for compensation. Although the compensation is, in principle, possible at the broadcasting station, such a compensation will generally not be complete, due to the dependency of the differences in transmission paths on the particular location of the moving station. Therefore, at least part of the compensation is always necessary to be performed at the receiving site. In view of the possibility of an initial compensation at the broadcasting station, the invention may also be applied to a case, wherein the first signal is terrestrially transmitted, and the second signal is transmitted via satellite.

[0024] Also preferably, the first and second audio broadcast signals respectively transmit one and the same audio service via different broadcast systems. Any broadcast environment, wherein one audio service is transmitted via different broadcast systems such as digital (for instance DAB or DRM) and analogue (for instance FM or AM) are called "simulcast transmissions" in the art.

[0025] Preferably, the downsampled versions of the sound signals that are employed for determining a delay therebetween and controlling the variable delay for compensation, are obtained by reducing a sampling frequency (also called sampling rate) thereof. Further preferably, a lowpass filter, such as a FIR (Finite Impulse Response) filter is applied to each of the sound signals before downsampling. Thereby, artefacts that may occur due to high frequency components in downsampling may be eliminated. Further preferably, the downsampling of each of the sound signals is performed in two stages, respectively. The first downsampling portion therefore comprises a first and a second decimation stage. In the first decimation stage, the first reproduced sound signal is downsampled with a first downsampling ratio, and the result thereof is further downsampled by the second decimation stage with a second downsampling ratio. In the same manner, the second downsampling portion comprises a third and a fourth decimation stage wherein the second reproduced sound signal is subsequently downsampled with a third and a fourth downsampling ratio. The downsampling ratio is the ratio of a sampling frequency of a signal before a particular downsampling stage to the sampling frequency after the respective downsampling stage.

[0026] A two-stage downsampling of the two audio signals is particularly advantageous, since FIR filters of low order can be used for average level detecting of signals to be downsampled, without running in problems with aliasing effects.

[0027] Preferably, the delay control portion determines an initial delay time of the second reproduced sound signal with respect to the first reproduced sound signal, in order to control the variable delay time of the variable delay portion so as to compensate for the initial delay time. Delay time compensation may be partially (delay reduction) or completely. Preferably, the initial delay is completely compensated by the variable delay of the first signal, within the limits of the calculation precision of the initial delay time. In accordance with the present invention, a precision of initial delay determination of 1 ms is possible.

[0028] Preferably, the delay control portion determines the initial delay time of the second reproduced sound signal with respect to the first reproduced sound signal by detecting the maximum of a cross-correlation function between the downsampled first and downsampled second reproduced sound signals. In order to reduce calculation power, cross-correlation is performed by convolution in the frequency domain. Cross-correlation in time domain would need a high calculation power for long vectors.

[0029] Preferably, after performing the cross calculation between the two signals, transmission gaps, muted audio at the transmitter side or transmitted sine-frequencies are detected in a subsequent silence detection step. Thereby, artefacts due to signal portions that are not usable for delay time detection by cross-correlation can be eliminated.

[0030] According to a preferred embodiment, the variable delay portion comprises a ring buffer with variable length. The variable delay time of the ring buffer can be controlled in correspondence with the initial delay time to be compensated, and indicated by a control signal issued by the control portion based on the cross-correlation function of the downsampled sound signals.

[0031] Preferably, downsampling is employed to obtain reproduced sound signals originating from the different transmission paths having only minimal differences in sampling frequencies. The invention can reduce differences in the sampling frequencies of the two signals much below 1%.

[0032] Further embodiments of the present invention are the subject matter of the dependent claims.

[0033] 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 illustrates the overall system architecture of an exemplary embodiment of an audio receiver according to the present invention,

Fig. 2 is a detailed block scheme of essential components for an exemplary audio receiver for implementing the present invention according to a particular embodiment, and

Fig. 3 is a flowchart illustrating steps of a method in accordance with the present invention.

[0034] Illustrative embodiments of the present invention will now be described with reference to the drawings.

[0035] The present invention provides a scheme enabling calculation time reduction for calculating a delay time between two time-shifted audio signals having different sampling frequencies such as a (digitalized) analogue broadcast signal and a digital broadcast signal. The calculation time reduction is achieved by downsampling the original audio signals to one similar sampling frequency. The downsampling is preferably done in two steps to reduce the needed calculation time further. "Similar sampling frequency" means that the relative amount of the difference in sampling frequencies between both signals with respect to the absolute amount of one of the sampling frequencies is considerably reduced after downsampling with respect to the original relative difference before downsampling. Experiments showed that a relative difference in the sampling frequency below 0.23% is acceptable.

[0036] Fig. 1 illustrates a general overview of the structure of an exemplary audio receiver according to the present invention. The audio receiver comprises a first antenna 10, a second antenna 20, a first signal reception portion 12 and a second signal reception portion 22, a first downsampling portion 16 and a second downsampling portion 26, a variable delay portion 18, a delay control portion 30, a selection portion 32 and a selection control portion 34.

[0037] The first and the second signal reception portions 12 and 22 receive and process audio broadcast signals 11 and 21 that come in via antennae 10 and 20, respectively. Preferably, the audio signals 11 and 21 transmit one and the same audio service (i.e. one and the same audic program) via two different broadcast systems (simulcast). For instance, the first received signal is an analogue broadcast signal such as FM, and the second received signal is a digital audio broadcast signal such as DAB. In case of simulcast transmission of analogue and digital signals, due to coding delays in the digital broadcast transmission chain, there is a considerable delay of the digital signal with respect to the analogue signal. The delay has to be compensated in order to enable a seamless and inaudible switching between the two signals. In the example of DAB and FM, the average delay is about 1.4 seconds. Generally, however, the delay may vary from station to station, and has, therefore, to be dynamically determined at the receiving apparatus during operation.

[0038] The processing performed by the first and second signal reception portions 12 and 22, respectively results in a reproduced sound signal that is obtained on the basis of the received audio broadcast signals. In case of receiving an analogue and a digital signal, the first and the second signal reception portions 12 and 22 are an analogue audio broadcast signal receiving portion and a digital audio broadcast signal receiving portion, respectively.

[0039] Typical processing in an analogue audio broadcast signal receiving portion includes the steps to tune in to an analogue audio broadcast signal, such as a FM audio broadcast signal 11, and further includes a demodulation processing and a de-emphasis processing. The reproduced analogue sound signal 14 is digitized and sampled with a first predetermined sampling frequency.

[0040] Typical processing in a digital audio broadcast signal receiving portion includes tuning in to a digital audio broadcast signal 20, audio and channel decoding, and time de-interleaving to produce a reproduced digital sound signal 24 constituted with time de-interleaved audio information data. For further processing, the reproduced digital sound signal 24 is sampled with a second predetermined sampling frequency.

[0041] The variable delay portion 18 is operable to delay the first reproduced sound signal 14 (in a preferred embodiment: obtained from an analogue broadcast signal 12) with a variable delay time. The output of variable delay portion 18 is a time delayed first reproduced sound signal 14'. Control of the variable delay portion 18, and, in particular, the variable delay time is achieved by delay control portion 30. Control of the variable delay time by the delay control portion 30 is performed based on a calculation of an initial delay time of the second reproduced sound signal 24 with respect to the first reproduced sound signal 14. In case of a simulcast transmission, the delay time is defined as the delay between a particular portion of the broadcast program in the sound signal 24 reproduced by the second signal reception portion 22 with respect to the same portion of the program in the first sound signal 14 as reproduced by the first signal reception portion 12. Alternatively, the delay can for instance be determined on the basis of predetermined pilot signals or time stamps included in both received broadcast signals. Such a determination is also suitable, if there is no simulcast situation.

[0042] Calculation of the initial delay time between two signals having different sampling rates is generally complicated and processing time consuming, in case of signals with two different sampling frequencies. According to the present invention, the calculation processing by the delay control portion 30 is simplified by downsampling both first reproduced sound signal 14 and second reproduced sound signal 24 by different downsampling ratios. Thereby, a high degree of coincidence between the sampling rates of the first and the second reproduced sound signal can be achieved, i.e. the relative difference between the sampling frequencies of the first and second downsampled sound signals may be reduced to a value of an order of 0.2% or lower. Downsampling is performed by first downsampling portion 16 and second downsampling portion 26, for the first and the second reproduced sound signal 14 and 24, respectively.

[0043] Selection unit 32 receives both the delayed first reproduced sound signal 14' and the second reproduced sound signal 24. Selection portion 32 operates to select one of the two input audio signals 14' and 24 and output a selected one of the signals for being perceived by the user. Selection portion 32 is controlled by selection control portion 34. Generally, selection control by the selection control unit 34 is performed in such a manner that the user is capable of always perceiving the signal currently having the best audio quality, even in case of a moving receiver such as in a vehicle. In the case when the quality of the currently selected audio signal decreases (for instance if the currently perceived audio signal is from digital broadcast, and the vehicle drives out of the service area of the digital broadcasting station), automatic switchover is performed. Since the timing of the delayed first reproduced sound signal 14' and the second reproduced sound signal 24 coincide to a high degree, switchover by the selection unit 32 can be performed seamlessly.

[0044] Fig. 2 is a more detailed view of a portion of the audio receiver in a particular embodiment. In the particular embodiment, it is assumed that the first reproduced sound signal (FM_In 214) is a digitized analogue FM signal with a sampling frequency of 44,100 Hz and the second reproduced sound signal (DAB_In 224) is a digital audio signal (DAB signal) with a sampling frequency of 48,000 Hz. In the embodiment, the variable delay portion is implemented by means of ring buffer 218 with variable length. Ring buffer 218 outputs a delayed sound signal with 44,100 Hz sampling rate. The variable time delay of the ring buffer 218 is controlled by a control signal 238 for setting of the ring buffer dimension issued by delay control portion 230. Delay control portion 230 operates by correlating the first and the second reproduced sound signal after having been downsampled in first downsampling portion 216 and second downsampling portion 226.

[0045] First downsampling portion 216 includes first decimation stage 216a and second decimation stage 216b. Decimation stage 216a includes a FIR lowpass filter 216b1 of 32nd order and downsampling section 216a2 for downsampling the received signal by a ratio of 1:11. Second decimation stage 216b includes 64th order FIR lowpass filter 216b1 and downsampling section 216b2 with a downsampling ratio of 1:4.

[0046] Second downsampling section 226 includes third decimation stage 226a and fourth decimation stage 226b. Third decimation stage 226a comprises 32nd order FIR lowpass filter 226a1 and downsampling section 226a2 having a downsampling ratio of 1:8. Fourth decimation stage 226b includes 64th order FIR lowpass filter 226b1 and downsampling section 226b2 having a downsampling ratio of 1:6.

[0047] Delay control portion 230 comprises two Fast Fourier Transform (FFT) sections 231 and 232. Further, delay control portion 230 comprises convolution section 233 and Inverse Fast Fourier Transform (IFFT) section 234. In the particular embodiment, control portion 230 further includes, besides peak detection section 236, a silence detection section 235.

[0048] The difference in sampling weights between the DAB audio signal and the digitized FM audio signal is due to the specification of digital FM demodulators available in the art. While DAB audio signals are digitally transmitted with a sampling frequency of 48,000 Hz (48 kHz), the available digital FM demodulators use a sampling frequency of 44,100 Hz (44.1 kHz), which is the sampling frequency used by the CD.

[0049] To perform the cross-correlation between the two signals, the sampling frequencies of the two signals have to be matched. This is achieved by a two-step downsampling procedure wherein two different downsampling ratios are applied by the first and second downsampling portions, respectively. The first decimation stage 216a receives the reproduced FM sound signal 214 with a sampling rate of 44.1 kHz. After downsampling the signal with a downsampling ratio of 1:11, the signal has an intermediate sampling frequency of approximately 4 kHz (exactly: 4009.9 Hz). Said signal is further downsampled in second decimation stage 216b with the downsampling ratio of 1:4. The result is a signal with a sampling frequency of approximately 1 kHz (exactly: 1,002.3 Hz).

[0050] Third decimation stage 226a receives reproduced DAB sound signal sampling frequency 48 kHz. After downsampling with a ratio of 1:8, the sampling frequency is 6 kHz (6,000 Hz). The downsampled signal is further downsampled in the fourth decimation stage 226b with a downsampling ratio of 1:6. The resulting signal of the fourth decimation stage 226b has a sampling rate of 1 kHz (1,000 kHz). Accordingly, the sampling rates after both signals have been downsampled in two stages with different downsampling ratios respectively, coincide to a high degree. The relative difference in sampling frequencies is as low as 0.23%. Experiments showed that such a small difference in the sampling frequencies is acceptable.

[0051] It is a major advantage of the present invention that both signals are converted to almost the same low sampling frequency of 1 kHz in software. Thereby, necessary calculation power to perform the cross-correlation is reduced considerably and at the same time a possible resolution of the determined delay time of 1 ms is possible. Such a high accuracy of delay time determination and compensation is sufficient to perform an inaudible switch between the two signals if so desired.

[0052] For determining the delay in the digitized FM and DAB audio streams, cross-correlation is used. Because the cross-correlation in the time domain needs a high calculation power for long vectors, the downsampled signals input to delay control portion 230 are first transformed into the frequency domain by FFT sections 231 and 232, respectively. Cross-correlation is performed by convolution in the frequency domain in convolution section 233. After correlation, correlation results undergo inverse FFT in IFFT section 234.

[0053] In the particular embodiment described with reference to Fig. 2, after performing the cross-correlation between the two signals, and re-transformation into the time domain, the resulting signal (cross-correlation function) is processed by silence detection section 235 to perform a "silence test" to ensure useful interpretation of the cross-correlation function. The silence detection section 235 detects transmission gaps, muted audio at the transmitter side or transmitted sine-frequencies. If such audio signals occur during the determination of the delay, the cross-correlation function has no clear maximum peak. Signals the cross-correlation function of which has no clear maximum peak are not usable for delay determination by peak detection. Therefore, silence detection section 235 determines if the maximum value of the cross-correlation function is much higher than the average value (i.e. a clear peak is present). If this is not the case, it is considered that the currently received signals are not usable for delay determination, and the whole process is started again with new samples of the audio signals. By processing of the silence detection section 235, delay control section 230 takes into account that the audio may be muted or otherwise deteriorated so that no clear peak in the cross-correlation function can be detected. In the case that the audio signal is muted, the delay detection must be repeated until the audio signal is not muted.

[0054] The subsequent last processing stage is performed by peak detection section 236. Peak detection section 236 calculates the delay time by detecting the peak (maximum) of the cross-correlation function. The position of the maximum of the cross-correlation function between the FM and the DAB audio signals determines the delay between the two signals, measured in the number of samples (after downsampling). The thus determined initial delay of the second signal with respect to the first signal determines the delay time of ring buffer 218. Ring buffer 218 shall delay the FM audio signal 214 to produce a delayed FM audio signal 214' that is delayed with respect to initial signal 214 by the delay time of signal 224 with respect to signal 214, as determined by control section 230 (delay time compensation). Thus, the calculated value of the initial delay time is taken to define the dimension of ring buffer 218 to appropriately delay FM audio signal 214 to compensate for the initial delay of digital signal 224. After the delay of FM signal 214, delayed FM audio signal 214' and DAB audio 224 are aligned, and a switch between the two signals can be performed seamlessly and inaudible to the user.

[0055] It has to be noted that the particular exemplary data, especially the indicated values concerning sampling frequencies, downsampling ratios, FIR low pass filter orders etc. are given by way of exemplary description only in the foregoing description. The invention is not limited to these particular values and respective modifications to other values, or employing, for instance, only a single decimation stage for downsampling each signal, are possible within the present invention. Fig. 3 shows an exemplary flowchart of a method for determining the control signal setting a variable delay time in accordance with the present invention.

[0056] The initial processing of steps S100 to S130 on the left-hand side of the flowchart of Fig. 3, and of steps S200 to S230 on the right-hand side of the flowchart are performed in parallel, for the analogue and digital received audio signals, respectively. In step S100 and respectively S200, an analogue and a digital audio signal are received. As indicated above, the specifics of the received signals is not limited to an analogue and a digital signal, but other kinds of signals such as two different kinds of digital signals, or audio broadcast signals distributed by satellite and terrestrially, respectively, are generally also possible.

[0057] In step S110, the received signal (in the example: analogue signal) is processed to obtain the digitized analogue sound signal with a first sampling frequency at step S120. A respective signal processing of the second received signal (in the example: digital signal) is performed in step S210 to obtain a reproduced digital sound signal with a second sampling frequency in step S220. Downsampling of the respective signals is performed by corresponding processing steps S130 for the analogue signal and S230 for the digital signal.

[0058] In subsequent step S300, cross-correlation is performed between the two downsampling signals. As described above, preferably, the cross-correlation is performed by convolution in the frequency domain.

[0059] Subsequent step S310 detects the position of the peak of the resulting cross-correlation function. In the preferred embodiment as described above, in an intermediate step (not shown in Fig. 3) between steps S300 and S310, silence detection is performed, in order to disregard any correlation result having no clear peak, and therefore not usable for delay determination. On the basis of the peak detected in step S31 0, the initial delay of the second (digital) audio signal with respect to the first (analogue) audio signal is determined. The determined initial delay forms the basis for generating a control signal in step S330 that determines the necessary amount of variable time delay to be applied by variable delay portion 18, in order to compensate for the detected initial delay. Step S340 instructs variable delay portion 18 by setting the respective variable delay time.

[0060] After the analogue signals have passed the variable delay portion, there is no noticeable delay between the analogue and the digital signal any more and switching between the two signals can be performed at any time.

[0061] 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 plural further modifications of the described embodiments.

[0062] In summary, the present invention relates to a particular efficient scheme for determining an initial delay of a second received audio signal with respect to a first received audio signal, specifically suitable for a simulcast environment. In order to receive a seamless and inaudible switchover in a simulcast environment, the initial delay time must be exactly determined and compensated by delaying the first signal at the receiving apparatus by a respective amount. Calculation power requirements are particularly high if the sound signals reproduced from first and second received audio signals have different sampling frequencies. The present invention adapts the sampling frequencies by downsampling both signals for delay time determination and thus considerably reduces calculation effort.

Claims

1. An audio receiver, comprising:

a first signal receiving portion (12) for receiving a first audio broadcast signal (10) to obtain a first reproduced sound signal (14, 214),
a second signal receiving portion (22) for receiving a second audio broadcast signal (20) to obtain a second reproduced sound signal (24,224),
a variable delay portion (18, 218) for delaying the first reproduced sound signal (14, 214) by a variable delay time, and
a delay control portion (30, 230) for controlling said variable delay portion (18, 218),
characterized by
a first downsampling portion (16, 216) for downsampling the first reproduced sound signal (14, 214),
a second downsampling portion (26, 226) for downsampling the second reproduced sound signal (24, 224), and in that
said delay control portion (30, 230) controlling said variable delay portion (18, 218) based on the first and second reproduced sound signals downsampled by said first and second downsampling portions.

2. An audio receiver according to claim 1, wherein said first audio broadcast signal (10) being an analogue signal, and said second audio broadcast signal (20) being a digital signal.

3. An audio receiver according to claim 1 or 2, wherein one of said audio broadcast signals (10, 20) being transmitted via satellite and the other of said audio broadcast signals (10, 20) being transmitted via terrestrial transmission.

4. An audio receiver according to any of claims 1 to 3, wherein said first and said second audio broadcast signals (10, 20) respectively transmit one audio service via different broadcast systems.

5. An audio receiver according to any of claims 1 to 4, wherein
said first downsampling portion (16, 216) comprising a first decimation stage (216a) for downsampling said first reproduced sound signal (14, 214) with a first downsampling ratio and a second decimation stage (216b) for downsampling the signal downsampled by the first decimation stage (216a) with a second downsampling ratio; and
said second downsampling portion (26, 226) comprising a third decimation stage (226a) for downsampling said second reproduced sound signal (24, 224) with a third downsampling ratio and a fourth decimation stage (226b) for downsampling the signal downsampled by the third decimation stage (226a) with a fourth downsampling ratio.

6. An audio receiver according to any of claims 1 to 5, wherein said delay control portion (30, 230) determining an initial delay time of the second reproduced sound signal (24, 224) with respect to the first reproduced sound signal (14, 214), in order to control the variable delay time of the variable delay portion (18, 218) so as to compensate for the initial delay time.

7. An audio receiver according to claim 6, wherein said delay control portion (30, 230) comprising a correlation section (231, 232, 233, 234) for determining said initial delay time of the second reproduced sound signal (24, 224) with respect to the first reproduced sound signal (14, 214) by detecting the maximum of a cross-correlation function between the downsampled first and downsampled second reproduced sound signals.

8. An audio receiver according to any of claims 1 to 7, wherein said variable delay portion (18) comprising a ring buffer (218) with variable length.

9. A method of receiving audio broadcast signals (10, 20), comprising the steps of
receiving (S100) a first audio broadcast signal (10),
obtaining (S110, S120) a first reproduced sound signal (14, 214) from said received first audio broadcast signal (10),
receiving (S200) a second audio broadcast signal (20),
obtaining (S210, S220) a second reproduced sound signal (24, 224) from said received second audio broadcast signal (20), and
delaying the first reproduced sound signal (14, 214) by a variable delay time, **characterized by** the steps of

downsampling (S130) the first reproduced sound signal (14),
downsampling (S230) the second reproduced sound signal (24), and
controlling (S300, S310, S320, S330, S340) said variable delay time based on the downsampled first and second reproduced sound signals.

10. A method according to claim 9, wherein said first audio broadcast signal (10) being an analogue signal, and said second audio broadcast signal (20) being a digital signal.

11. A method according to claim 9 or 10, wherein one of said audio broadcast signals (10, 20) being transmitted via satellite and the other of said audio broadcast signals being transmitted via terrestrial transmission.

12. A method according to any of claims 9 to 11, wherein said first and said second audio broadcast signals (10, 20) respectively transmit one audio service via different broadcast systems.

13. A method according to any of claims 9 to 12, wherein each of said downsampling steps (S130, S230) is performed in two stages, such that said first reproduced sound signal (14, 214) is subsequently downsampled with a first downsampling ratio and a second downsampling ratio, and said second reproduced sound signal (24, 224) is subsequently downsampled with a third downsampling ratio and a fourth downsampling ratio.

14. A method according to any of claims 9 to 13, wherein said controlling step comprising the step (S310, S320) of determining an initial delay time of the second reproduced sound signal (24, 224) with respect to the first reproduced sound signal (14, 214), in order to control the variable delay time so as to compensate for the initial delay time.

15. An audio receiver according to claim 14, wherein said determining step comprising the step of detecting (S310) the maximum of a cross-correlation function between the downsampled first and downsampled second reproduced sound signals.

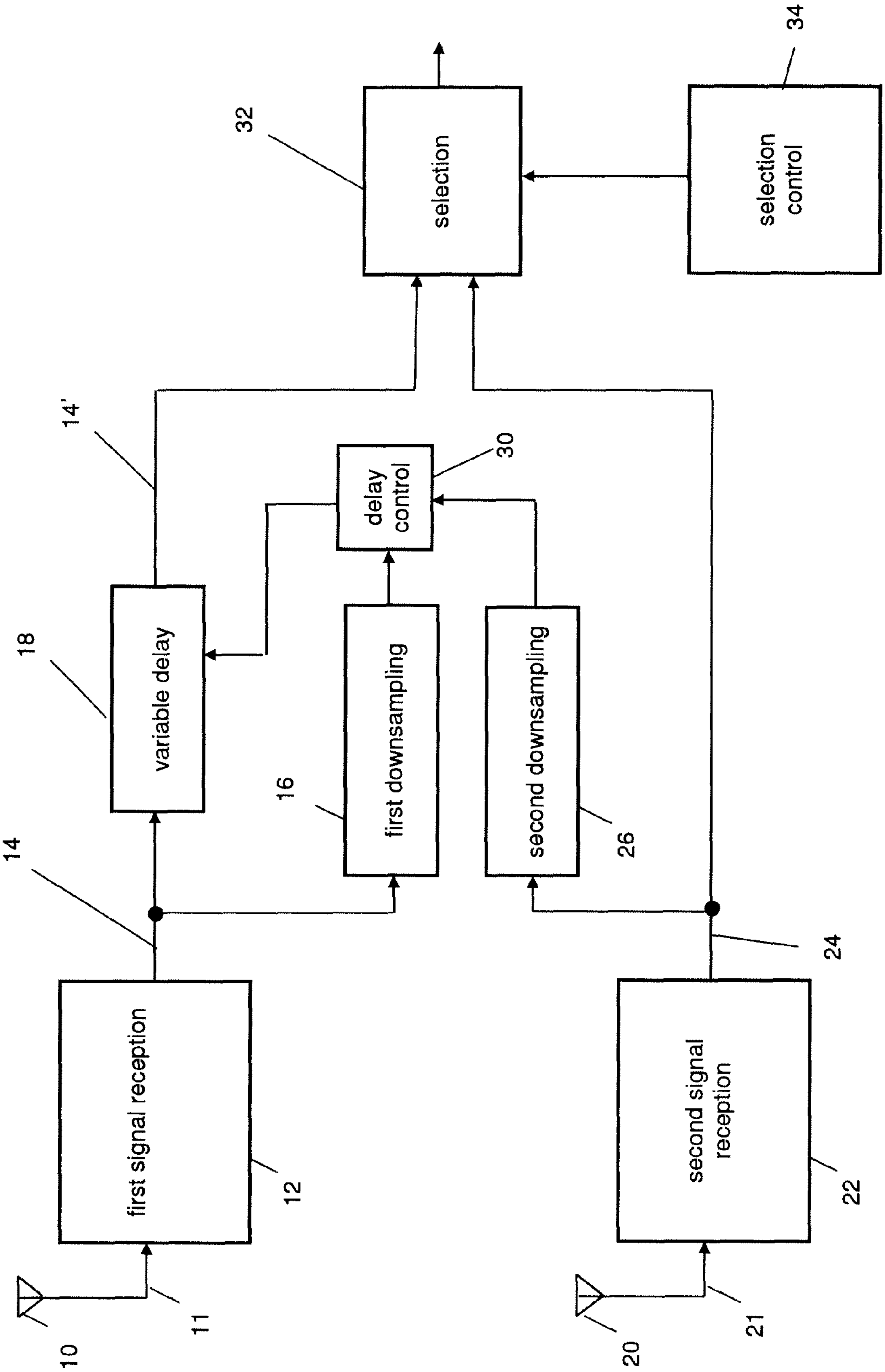
first signal reception
second signal reception
variable delay
first downsampling
second downsampling
delay control
selection
selection control
10
11
12
14
14'
16
18
20
21
22
24
26
30
32
34

Fig. 1

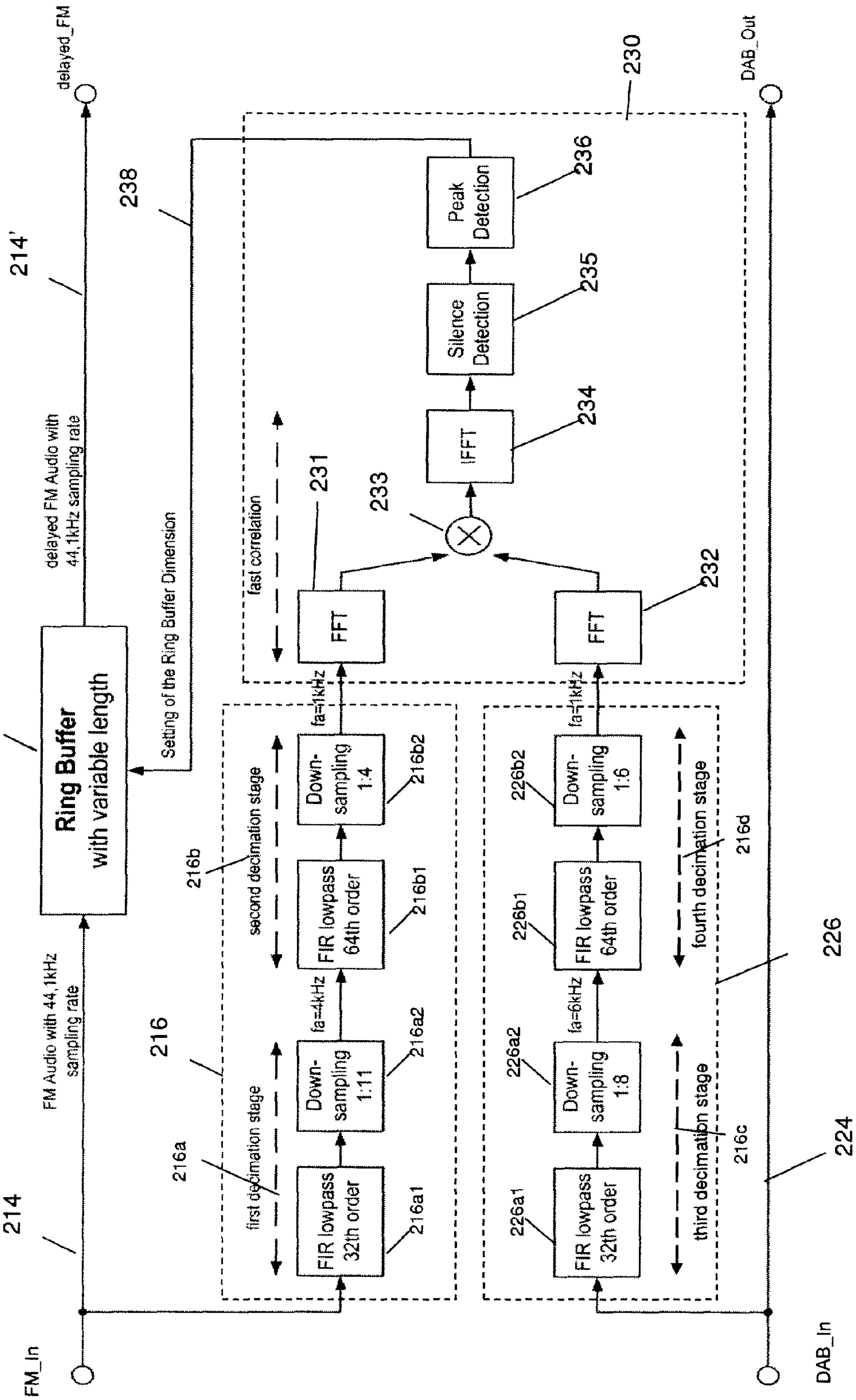
FM_In
214
FM Audio with 44,1kHz sampling rate
218
Ring Buffer
with variable length
delayed FM Audio with 44,1kHz sampling rate
214'
delayed_FM
Setting of the Ring Buffer Dimension
238
216
216a
first decimation stage
FIR lowpass 32th order
216a1
Down-sampling 1:11
216a2
fa=4kHz
216b
second decimation stage
FIR lowpass 64th order
216b1
Down-sampling 1:4
216b2
fa=1kHz
230
fast correlation
FFT
231
233
IFFT
234
Silence Detection
235
Peak Detection
236
FFT
232
fa=1kHz
226
226a1
FIR lowpass 32th order
226a2
Down-sampling 1:8
216c
third decimation stage
fa=6kHz
226b1
FIR lowpass 64th order
226b2
Down-sampling 1:6
216d
fourth decimation stage
224
DAB_In
DAB_Out

Fig. 2

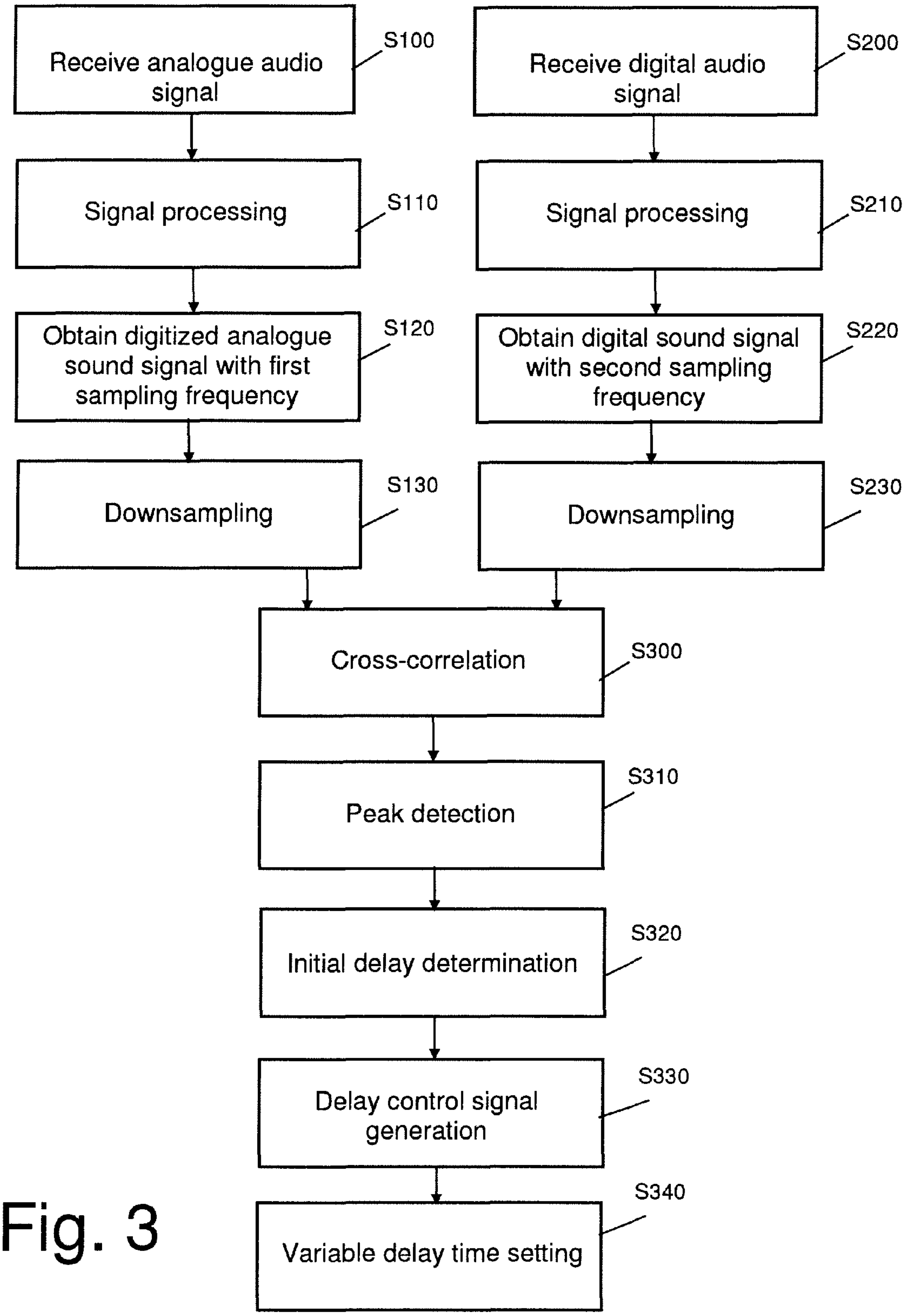
Receive analogue audio signal
S100
Signal processing
S110
Obtain digitized analogue sound signal with first sampling frequency
S120
Downsampling
S130
Receive digital audio signal
S200
Signal processing
S210
Obtain digital sound signal with second sampling frequency
S220
Downsampling
S230
Cross-correlation
S300
Peak detection
S310
Initial delay determination
S320
Delay control signal generation
S330
Variable delay time setting
S340

Fig. 3

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EUROPEAN SEARCH REPORT

Application Number
EP 11 16 9843

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
X	EP 1 233 556 A1 (SONY INT EUROPE GMBH [DE]) 21 August 2002 (2002-08-21) * paragraphs [0003] - [0004], [0008], [0014] - [0015], [0017], [0021] - [0022], [0030] * * figure 1 * -----	1-15	INV. H04H20/26 H04H60/11 H04H20/22
A	RICHARD C CABOT: "A Note on the Application of the Hilbert Transform to Time Delay Estimation", IEEE TRANSACTIONS ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, IEEE INC. NEW YORK, USA, vol. ASSP-29, no. 3, 1 June 1981 (1981-06-01), pages 607-609, XP007910107, ISSN: 0096-3518, DOI: DOI:10.1109/TASSP.1981.1163564 * "Introduction" chapter, lines 1-5 * -----	7,15	

TECHNICAL FIELDS SEARCHED (IPC)
H04H G10L

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
The Hague	19 July 2011	Iovescu, Vladimir

2

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19-07-2011

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		US 2002115418 A1	22-08-2002

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

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- EP 1227608 A2 **[0012]**