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(54) **AUDIO PROCESSOR**

(57) A method and apparatus of audio processing for a digital radio is described. A primary audio signal including music is received; a secondary audio signal is generated comprising a delayed version of the primary audio signal, the delay being determined from a characteristic

of the music; an audio output signal is generated by cross-fading between the primary audio signal and the secondary audio signal in response to a change in audio error status of the primary audio signal.

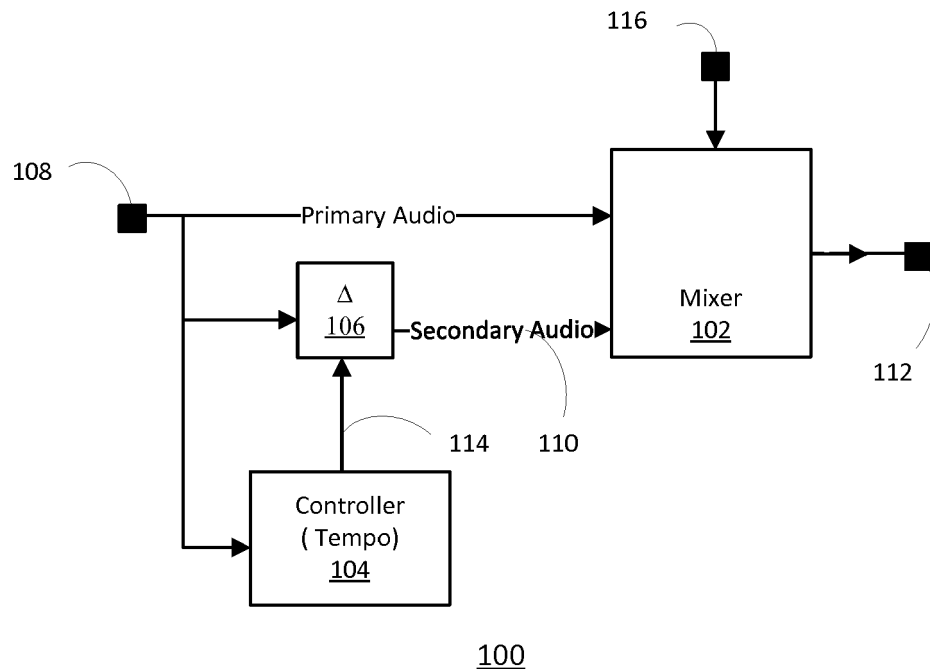


FIGURE 1

Description

[0001] This disclosure relates to audio processors suitable for digital radio.

[0002] In digital radio broadcasts, audio signals are encoded in the digital domain. The digital radio broadcast is transmitted over error-prone channels, due to which the received (encoded) signals may contain bit errors. The number of bit errors typically increases as the reception quality deteriorates. If the bit errors are still present after all error detection and error correction methods have been applied, the corresponding audio frame may no longer be decodable and is corrupted either completely or in part. One way of dealing with these errors is to mute the audio output for a certain period of time, for example during one or more frames. In other approaches, the corrupted signal sections are detected, after which they are replaced by silence (muting), by signal sections from the same channel (repetition) or from an adjacent channel (left/right substitution). The signal sections may be replaced completely or only one or several frequency bands may be replaced. An additional approach is that of noise substitution, where an audio frame may be replaced by a noise frame, the spectral envelope of which may be matched to that expected from the audio frame.

[0003] Various aspects of the disclosure are defined in the accompanying claims. In a first aspect there is described an audio processor for a digital radio comprising: an audio input configured to receive a primary audio signal, the primary audio signal comprising music; an audio output for an audio output signal; an audio error status input configured to receive an audio error status; wherein the audio processor is configured to generate a secondary audio signal by delaying at least one of the primary audio signal and the audio output, and to generate an audio output signal by cross-fading between the primary audio signal and the secondary audio signal in response to a change of the audio error status, wherein the delay is determined from a characteristic of the music.

[0004] In one or more embodiments, the characteristic of the music may comprise the tempo and wherein the secondary audio signal is beat-aligned with the primary audio signal.

[0005] In one or more embodiments, the characteristic of the music may comprise the time signature and wherein the secondary audio signal is bar-aligned with the primary audio signal.

[0006] In one or more embodiments, the audio processor may comprise a mixer, the mixer comprising a first mixer input coupled to the audio input, a second mixer input, and a mixer output coupled to the audio output, and wherein the mixer is configured to receive a secondary audio signal on the second mixer input and, in response to a change in the error status, to cross-fade between the primary audio signal and the secondary audio signal by modifying the primary audio signal with a primary mixing factor, modifying the secondary audio signal with a secondary mixing factor and combining the modified primary audio signal and the modified secondary audio signal.

[0007] In one or more embodiments, the audio processor may comprise a time delay module having a delay input and a delay output, wherein the delay input is coupled to the audio input and wherein the time delay output is coupled to the second mixer input.

[0008] In one or more embodiments, the audio processor may comprise a time delay module have a delay input and a delay output, wherein the delay input is coupled to the audio output and wherein the time delay module is configured to output the secondary audio signal.

[0009] In one or more embodiments, the audio processor may comprise a time delay control module having an output coupled to the time delay module and an input coupled to the audio input, wherein the control module is configured to vary the delay of the time delay module dependent on the characteristic of the music.

[0010] In one or more embodiments, the audio processor may comprise at least one of a loudness level aligner and spectrum aligner having a respective first input coupled to the primary audio input and a respective second input coupled to the delay module output.

[0011] In one or more embodiments of the audio processor, the mixer may be configured to generate a first mixing factor and a second mixing factor, and the audio mixer is configured to apply the first mixing factor to the primary audio signal and to apply the second mixing factor to the secondary audio signal, and to combine the two signals; wherein in response to the audio error status indicating an error, the first mixing factor changes from a 1 value to 0 value and the second mixing factor changes from a 0 value to a 1 value, and in response to the audio error status indicating no error, the first mixing factor changes from a 0 value to a 1 value and the second mixing factor changes from a 1 value to a 0 value.

[0012] In one or more embodiments, the audio processor may comprise an audio decoder configured to receive an audio signal and output the primary audio signal and the audio error status.

[0013] In a second aspect there is described a method of audio processing for a digital radio, the method comprising receiving a primary audio signal comprising music; receiving an audio error status of the primary audio signal; generating a secondary audio signal comprising a delayed version of the primary audio signal, the delay being determined from a characteristic of the music; and generating an audio output signal by cross-fading between the primary audio signal and the secondary audio signal in response to a change in audio error status.

[0014] In one or more embodiments, generating the secondary audio signal may further comprise applying level alignment to substantially align the level of the secondary audio signal with the primary audio signal.

[0015] In one or more embodiments, generating the secondary audio signal may further comprise applying spectrum

alignment to substantially align the spectrum of the secondary audio signal with the primary audio signal.

[0016] In one or more embodiments, generating the secondary audio signal may comprise applying a time delay to the primary audio signal.

[0017] In one or more embodiments, generating the secondary audio signal may comprise applying a time delay to the audio output signal.

[0018] In a third aspect there is described a computer program product comprising instructions which, when being executed by a processing unit, cause said processing unit to perform the steps of receiving a primary audio signal comprising music; receiving an audio error status of the primary audio signal; generating a secondary audio signal comprising a delayed version of the primary audio signal, the delay being determined from a characteristic of the music; and generating an audio output signal by cross-fading between the primary audio signal and the secondary audio signal in response to a change in audio error status.

[0019] In the figures and description like reference numerals refer to like features. Embodiments of the invention are now described in detail, by way of example only, illustrated by the accompanying drawings in which:

Figure 1 shows an audio processor according to an embodiment.

Figure 2 shows an example timing diagram of the error status signal, the primary mixing factor and the secondary mixing factor for the audio processor of figure 1.

Figure 3 shows an audio processor according to an embodiment.

Figure 4 shows an audio processor according to an embodiment.

Figure 5 shows an audio processor according to an embodiment.

Figure 6 shows a method of audio processing, according to an embodiment.

[0020] Figure 1 shows an audio processor 100 according to an embodiment. The audio processor 100 includes an audio mixer 102, a delay module controller 104 and a delay module 106. A primary audio input 108 for receiving an audio signal including music may be connected to a primary mixer input of the mixer 102. An audio output 112 of the audio processor 100 may be connected to an output of the mixer 102. A mixer control input 116 may be connected to a control output of an audio source decoder (not shown). The delay module 106 may have a delay module input connected to the audio input 108. A delay module output may be connected to a secondary mixer input 110. The delay module controller 104 may have an input connected to the primary audio input 108 and an output 114 connected to a control input of the delay module 106.

[0021] In operation, an audio input signal on audio input 108, which may be referred to as the primary audio signal is fed into the primary input of the mixer 102. The primary audio signal may be received from the output of an audio source decoder (not shown). The audio decoder may include one or more error detection and correction mechanisms. The primary audio signal may also be input to the delay module 106, the delay of which may be set by the control module 104. The audio signal output from the delay module is fed into the secondary input 110 of the mixer 102. The mixer 102 may switch or transition the output from the primary audio signal on primary input 108 to the secondary audio signal on secondary input 110 (and back again), controlled by a control signal on control input 116. The control signal may for example be an error status signal from an audio decoder (not shown). The output of the mixer 102 may output audio signal on the audio output 112.

[0022] The control signal on the mixer control input 116 may control the mixer to cross-fade from the primary audio signal to secondary audio signal when the error status indicates that a corrupted frame is present or will be present in a certain time. The control signal may further control the mixer 102 to cross-fade from the secondary audio signal to the primary audio signal on the primary audio 108 when the primary audio signal is no longer corrupted.

[0023] The controller 104 may receive the audio signal and adjust the delay of delay module 106 according to a characteristic of the music expressed for example as the beats-per-minute or the number of beats per bar. In one example, the number of beats per bar may be four, as this will align down-beats in many music genres, and therefore lead to bar-aligned audio inputs. The number of beats per bar may be unity, which will lead to beat-aligned audio inputs. It will be appreciated that the number of beats per bar may be derived by determining the time signature of the music. The number of beats per minute corresponds to the tempo of the music. The time corresponding to a single beat and the number of beats per bar may be determined by the control module 104 using a known audio tempo estimation method derived from the audio input. The time corresponding to a single beat may also be determined from other sources of information. For example, the time may be derived from meta-data that is contained in the digital bit-stream, which is decoded by the audio source decoder.

[0024] The operation of the audio processor 100 may be further understood with reference to figure 2. Figure 2 shows a timing diagram 150 of the error status signal 152, the primary mixing factor 154 which is denoted g_1 , and the secondary mixing factor 156 which is denoted g_2 . The error status value on mixer control input 116 may indicate which of the two audio inputs 108, 110 is the required output. In this example the error status value is denoted as a logic "1" to select the primary audio input 108 corresponding to no error detected, and a logic "0" to select the secondary audio input 110 indicating an error has been detected. It will be appreciated that the logic values may be different for other examples. In some examples the error status may include more than a single bit value which may then be decoded to control the mixer 102. The error status value may indicate whether the current audio frame is corrupted or not. In some examples the error status value may be received from a look-ahead feature of an audio decoder (not shown) indicating a future audio frame is corrupted. In this example, the error status may give advanced warning of a corrupted audio frame. The audio output signal s_{out} on audio output 112 may be generated by the mixer 102 from a combination of the primary input signal s_1 on the primary audio input 108 and the secondary input signal s_2 on the secondary audio input 110:

$$s_{out}=g_1 \cdot s_1 + g_2 \cdot s_2, \quad (1)$$

where g_1 and g_2 are mixing factors for the primary and secondary input signals generated by a mix factor generator in the mixer (not shown). The mixing factor generator may derive the mixing factor g_2 for the secondary audio input 110 from the mixing factor g_1 for the primary audio input 108 according to equation 2:

$$g_2=(1-g_1) \quad (2)$$

When the control signal 152 transitions from logic 1 to 0, indicating a corrupted audio frame, the mixing factor g_1 for the primary input signal shown as line 154 may transition from scalar 1 to 0, and correspondingly, the mixing factor g_2 for the secondary signal shown as line 156 may transition from scalar 0 to 1. As a result, the output audio signal, s_{out} , may smoothly cross-fade from the primary audio signal, s_1 , to the secondary audio signal, s_2 .

[0025] When the control signal 152 transitions from logic 0 to 1, indicating that the audio frame is valid again, the mixing factor g_1 for the primary input signal shown as line 154 may transition from scalar 0 to 1, and correspondingly, the mixing factor for the secondary signal shown as line 156 may transition from scalar 1 to 0. The output audio signal, s_{out} , will smoothly cross-fade from the secondary audio signal, s_2 , to the primary audio signal, s_1 .

[0026] The secondary audio signal may be beat-aligned or bar-aligned to the primary audio signal. The mixer module 102 may substitute the corrupted audio frame by a substitution audio segment that was received an integer number of beats beforehand.

[0027] In digital radio systems, audio decoding errors may occur due to varying reception conditions which result in a variation in the received RF signal strength by the digital radio.

[0028] When an audio frame is no longer decodable, known audio error concealment mechanisms often lead to audible and annoying artefacts, which break the continuity of the audio signal, especially in the case of music. When a corrupted frame is replaced by a noise frame, this is often perceived as artificial, even when the spectral properties of the noise are matched to that of the audio signal (when present).

[0029] The audio processor 100 may cross-fade the substitution audio segment from the secondary audio input 110 which may be beat-or bar-aligned to the missing audio segment. Since the substitution audio segment is derived from the same audio source, the substituted audio segment is likely to blend in more naturally with reduced perceived artefacts. Consequently, a person listening to the audio may perceive improved sound quality with fewer audio artefacts caused by audio decoding errors.

[0030] Figure 3 shows an audio processor 200 according to an embodiment. The audio processor 200 includes a mixer 202, a delay module controller 204 and a delay module 206. An audio input 208 for receiving an audio signal including music may be connected to a primary mixer input of the mixer 202. An audio processor output 212 of the audio processor 200 may be connected to an output of the mixer 202. A mixer control input 216 may be connected to a control output of an audio source decoder (not shown). The delay module 206 may have a delay module input connected to the audio output 212. A delay module output may be connected to a secondary mixer input 210. The delay module controller 204 may have an audio meta-data input 222 and an output 214 connected to a control input of the delay module 206.

[0031] In operation, a primary audio signal on audio input 208 may be received by the primary input of the mixer 202. The primary audio input signal may be generated from an audio source decoder (not shown) which includes one or more error detection and correction mechanisms. The audio signal at the audio processor output 212 may be input to the delay module 206, the delay of which may be set by the control module 204. The output of the delay module 206 is fed

into the secondary input 210 of the mixer 202. The mixer 202 may switch or transition the output from the primary audio signal on primary input 208 to the secondary audio signal on secondary input 210 (and back again), controlled by a control signal on control input 216. The control signal may for example be an error status signal from an audio decoder. The output of the mixer 202 may output the audio signal on the audio output 212.

[0032] The control or audio error status signal on the mixer control input 216 may control the mixer 202 to transition from primary audio signal to secondary audio signal by cross-fading between the two audio signals when the error status indicates that a corrupted frame is present or will be present in a certain time. The control signal further controls the mixer 202 to transition from the secondary audio signal on the secondary audio input 210 to the primary audio signal on the primary audio input 208 by cross fading between the two signals when the audio signal becomes available after a corrupted frame.

[0033] The delay module controller 204 may receive the audio signal and adjust the delay of delay module 206 according to the tempo of the music corresponding to the number of beats per minute and the time signature of the music corresponding to a number of beats per bar. The tempo and time signature of the music may be derived from meta-data that is contained in the digital bit-stream, which is decoded by the audio source decoder (not shown). The meta-data from the audio decoder may be received on the meta-data input 222.

[0034] By generating the secondary audio signal from a delayed version of the primary audio signal, the audio processor 200 may prevent corrupted audio frames being substituted by another corrupted frame that occurs several beats earlier for example four beats earlier.

[0035] For the audio processor 100 if the time between two corrupted frames is exactly that of the delay module, the substitution signal will contain invalid data at the time of the cross-fade, and therefore the mixer will output a corrupted audio frame. For audio processor 200, the first corrupted frame of audio data may be replaced by an audio frame from several beats earlier. This same audio frame may again be used during the second corrupted audio frame. Hence, the output of the mixer during the second corrupted audio frame will contain valid audio data rather than a corrupted signal.

[0036] Figure 4 shows an audio processor 250 according to an embodiment. The audio processor 250 includes an audio mixer 252, a loudness level aligner 272, a spectrum aligner 274, a delay module controller 254, and a delay module 256. The spectrum aligner 274 may include a filter-bank 270 and a level aligner 268. An audio input 258 for receiving an audio signal including music may be connected to a primary mixer input of the mixer 252. An audio processor output 262 of the audio processor 250 may be connected to an output of the mixer 252. A mixer control input 266 may be connected to the mixer 252. The delay module 256 may have a delay module input connected to the audio input 258. A delay module output 260 may be connected to a loudness level aligner input. The delay module controller 254 may have an input connected to the audio input 268 and an output 264 connected to a control input of the delay module 256.

[0037] The primary audio input 258 may be connected to the loudness level aligner 272, the filter-bank 270 the level aligner 268 and the mixer 252. The delay output 260 may be connected to the loudness level aligner 272. The output 276 of the loudness level aligner 272 may be connected to the input of the filter bank 270 of the spectrum aligner 274. The filter-bank output 278 may be connected to the level aligner 268. The output of the level aligner 268 may be connected to the secondary audio input 284 of the mixer 252.

[0038] In operation, a primary audio input signal on audio input 258 is fed into the primary audio input of the mixer 252, the loudness level aligner 272 and the spectrum aligner 274. The primary audio signal may also be input to the delay module 256, the delay of which may be set by the delay control module 254. The audio signal output from the delay module 256 is fed into the level aligner 272.

[0039] The loudness level aligner 272 may align the loudness level of the secondary audio signal to the primary audio signal by substantially equalising the loudness of the secondary audio signal to that of the primary audio signal. Equalising the loudness may improve the quality of the output signal, especially in the case where the music is considerably different across beats. The skilled person will appreciate that the loudness of the primary and secondary audio signals can be estimated by smoothing the signal power when the audio content is considered valid i.e. not corrupted. The level aligned secondary audio signal on level aligner output 276 may be received by the spectrum aligner 274.

[0040] The spectral aligner 274 may perform a spectral alignment operation, which substantially equalises the spectral content of the secondary audio signal to that of the primary audio signal. This may be done for example by splitting the primary and secondary audio signals into frequency bands using the filter-bank 270. The level aligner 268 may performing a level alignment for each frequency band. In other examples a frequency domain transform may be used to split the primary and secondary audio signals into frequency bands instead of the filter-bank 270.

[0041] Any delay caused by the processing performed for level-alignment and/or spectrum-alignment by the level aligner 272 and the spectrum aligner 274 may be subtracted from the delay of the delay module 256 such that the primary audio signal is synchronised with the secondary audio signal before mixing by the mixer 252.

[0042] It will be appreciated that the equalisation performed by the loudness level aligner 272 and the spectrum aligner 274 will not precisely align the audio characteristics of the primary and secondary audio signals but that they will be substantially the same. In some examples, the audio characteristics of the primary and secondary audio signals may be aligned within 2 dB.

[0043] In other examples of the audio processor, the loudness level aligner 272 and/or the spectrum aligner 274 may be omitted.

[0044] The mixer 252 may mix the primary audio signal on primary audio input 258 with the secondary audio signal on secondary audio input 284 following level and spectral alignment. The relative proportions of the primary audio signal and the secondary audio signal may be adjusted in response to the error status signal received on the mixer control input 266.

[0045] The mixer 252 may cross-fade the audio output from the primary audio signal on primary input 258 to the secondary audio signal on secondary input 284 (and back again), controlled by the error status signal on the mixer control input 266. The error signal may for example be an audio frame error signal from an audio decoder (not shown). The mixer 252 may output an audio signal on the audio processor output 262. The error signal may also indicate that error concealment has been used for the generation of the frame, due to which artefacts may be present on the audio signal in that frame.

[0046] The secondary audio signal present on secondary mixer input 284 may be bar-aligned or beat-aligned to the primary audio signal by controlling the delay between the primary audio signal and the secondary audio signal using the delay module 256 controlled by the delay module controller 254. The delay module controller 254 may set the delay of the delay module 256 in such a way that the delay between the primary audio signal and the secondary audio signal corresponds to an integer number of beats in the music signal. The number of beats may be four, as this will align down-beats in most music genres, and therefore leading to bar-aligned audio inputs. The number of beats may be unity, which will lead to beat-aligned audio inputs. The time corresponding to a single beat and/or the number of beats per bar may be determined from the primary audio signal in the tempo control module 254 using a typical audio tempo estimation method.

[0047] Example tempo estimation methods typically consist of two stages: a first stage that extracts features from the audio signal such as the envelope of the signal or detected onsets such as note changes, harmonic changes or percussive events or other salient changes in the music, and a second stage that estimates the periodicity of the features, for example by determining autocorrelation of the signal.

[0048] The mixer module 252 may substitute the corrupted audio frame by a substitution audio segment that was taken an integer number of beats before by cross-fading between the primary and secondary audio signals. Since the substitution audio segment is derived from the same audio source, the substituted audio segment is likely to blend in more naturally with reduced perceived artefacts. Consequently, a person listening to the audio processed by audio processor 250 may perceive improved sound quality with fewer audio artefacts because of audio decoding errors.

[0049] Figure 5 shows an audio processor 300 according to an embodiment. The audio processor 300 includes audio decoder 326 a mixer 302, a loudness level aligner 322, a spectrum aligner 324, a delay module controller 304, and a delay module 306. An audio input 318 for receiving a bit-stream representing a digitally encoded audio signal including music from an RF front-end (not shown) may be connected to audio decoder 326. An audio output 308 of the audio decoder 326 may be connected to a primary input of the mixer 302. An audio processor output 312 of the audio processor 300 may be connected to an output of the mixer 302. The audio processor output 312 may be connected to an input of delay module 306. The audio decoder 326 may have an error status output connected to the mixer control input 328. A delay module output 310 may be connected to an input of the loudness level aligner 322. The delay module controller 304 may have an input connected to the audio decoder meta-data output 328 and an output 314 connected to a control input of the delay module 306.

[0050] The audio output 308 of the audio decoder 326 may be connected to primary audio inputs of the loudness level aligner 322, the spectrum aligner 324, and the mixer 302. The loudness level aligner output 320 may be connected to the input of the spectrum aligner 324. The spectrum aligner output may be connected to the secondary input 332 of the mixer 302. In some examples, the spectrum aligner 324 and/or the loudness level aligner 322 may be omitted.

[0051] In operation, the audio signal on the audio decode output 308, which may be referred to as the primary audio signal, is received by the primary audio input of the mixer 302, the level aligner 322 and the spectrum aligner 324. The audio signal output from the mixer 302 may be input to the delay module 306, the delay of which may be set by the control module 304. The audio signal output from the delay module 306 is fed into the level aligner 322.

[0052] The level aligner 322 may align the loudness level of the secondary audio signal to that of the primary audio signal by equalising the loudness of the secondary audio signal to that of the primary audio signal. Equalising the loudness may improve the quality of the output signal, especially in the case where the music is considerably different across beats. The skilled person will appreciate that the loudness of the primary and secondary audio signals can be estimated by smoothing the signal power when the audio content is considered valid. The level aligned secondary audio signal may be received by the spectrum aligner 324.

[0053] The spectral aligner 324 may perform a spectral alignment operation, which equalises the spectral content of the secondary audio signal to that of the primary audio signal. In some examples the spectrum aligner may split the primary and secondary audio signals into frequency bands using a filter-bank. In other examples a frequency domain transform may be used to split the primary and secondary audio signals into frequency bands instead of a filter-bank.

[0054] The mixer 302 may mix the primary audio signal on primary audio input with the level and spectral aligned secondary audio signal on the secondary audio input 332. The relative proportions of the primary audio signal and the secondary audio signal may be controlled in response to the error status signal received on the mixer control input 316.

[0055] The mixer 302 may generate an audio output signal by cross fading from the primary audio signal received from the audio decoder output 308 to the secondary signal on secondary input 332 (and back again). The cross fading may be controlled by the error status signal on the mixer control input 316.

[0056] The error signal generated by the audio decoder 326 may be a look-ahead error signal, indicating an error on an audio frame. The audio frame may be buffered in the audio decoder for output at a later time to the primary audio input 308. In another example, a look-ahead error signal may be generated by the audio decoder 326 by performing a bit-stream analysis to detect an error. An error may be detected for example by decoding the header of an audio frame received by the audio decoder of a few frames ahead and performing a cyclic-redundancy-check or parity check before decoding the full audio signal on the current frame. The error signal generated in this way allows time for the mixer 302 to cross-fade to the secondary audio signal before the corrupted primary audio signal is present on the primary audio input 308. The output of the mixer 302 may output an audio signal on the audio output 312.

[0057] The secondary audio signal on secondary mixer input 332 may be bar-aligned or beat-aligned to the primary audio signal received from the audio decoder output 308 by controlling the delay of the delay module 306. The delay module controller 304 may set the delay of the delay module 306 in such a way that the delay between the primary audio signal and the aligned secondary audio signal corresponds to an integer number of beats in the music signal. The number of beats may be four, as this will align down-beats in most music genres, and therefore leading to bar-aligned audio inputs. The number of beats may be unity, which will lead to beat-aligned audio inputs. The delay may be derived by determining the time signature of the music. Information on the number of beats per minute and the time signature may be provided from metadata included in the audio signal, decoded by the audio decoder 326 and provided to the delay module controller 304 via audio decoder meta data output 328.

[0058] The mixer module 302 may substitute the corrupted audio frame on the primary audio signal by a substitution audio segment from the secondary audio signal that was received an integer number of beats earlier than the current audio segment. Since the substitution audio segment is derived from the same audio source, the substituted audio segment is likely to blend in more naturally with reduced perceived artefacts. Consequently, a person listening to the audio processed by audio processor 300 may perceive improved sound quality with fewer audio artefacts because of audio decoding errors.

[0059] By generating the substituted secondary audio signal from the audio processor output 312, the audio processor may reduce the number of invalid substitutions so reducing the period of time for which the audio is muted.

[0060] Figure 6 shows a method of audio processing for a digital radio 350. In step 352 a primary audio signal may be received comprising music. In step 354 a delayed and optionally level-aligned and/or spectrum-aligned version of the primary audio signal is generated, which may be referred to as a secondary audio signal. The secondary audio signal may be bar-aligned or beat-aligned with the primary audio signal. In step 356 the method may cross-fade between the primary audio signal and the aligned secondary audio signal in response to a change in error status of the primary audio signal.

[0061] An audio processing system for error concealment is described which may be included in a digital radio. The audio processing system may delay the audio signal by a time that corresponds to an integer number of beats in the music, and use the delayed signal as a substitution audio signal when the original audio is corrupted. When a corrupted audio frame is expected, the system cross-fades from the original audio to the substitution audio signal. The audio processing system cross-fades back to the original audio when the audio contains valid data again. The audio processor may be included in a digital radio. It will be appreciated that in other examples the audio processor may be included in any digital audio processing system.

[0062] A method and apparatus of audio processing for a digital radio is described. A primary audio signal including music is received; a secondary audio signal is generated comprising a delayed version of the primary audio signal, the delay being determined from a characteristic of the music; an audio output signal is generated by cross-fading between the primary audio signal and the secondary audio signal in response to a change in audio error status of the primary audio signal.

[0063] Embodiments of the audio processor may be implemented in hardware, software or a combination of hardware and software.

[0064] Although the appended claims are directed to particular combinations of features, it should be understood that the scope of the disclosure of the present invention also includes any novel feature or any novel combination of features disclosed herein either explicitly or implicitly or any generalisation thereof, whether or not it relates to the same invention as presently claimed in any claim and whether or not it mitigates any or all of the same technical problems as does the present invention.

[0065] Features which are described in the context of separate embodiments may also be provided in combination in a single embodiment. Conversely, various features which are, for brevity, described in the context of a single embodiment,

may also be provided separately or in any suitable sub combination.

[0066] The applicant hereby gives notice that new claims may be formulated to such features and/or combinations of such features during the prosecution of the present application or of any further application derived therefrom.

[0067] For the sake of completeness, it is also stated that the term "comprising" does not exclude other elements or steps, the term "a" or "an" does not exclude a plurality, a single processor or other unit may fulfil the functions of several means recited in the claims and reference signs in the claims shall not be construed as limiting the scope of the claims.

Claims

1. An audio processor for a digital radio comprising:

an audio input configured to receive a primary audio signal, the primary audio signal comprising music;
an audio output configured to output an audio output signal;
an audio error status input configured to receive an audio error status;
wherein the audio processor is configured to generate a secondary audio signal by delaying at least one of the primary audio signal and the audio output signal, and to generate the audio output signal by cross-fading between the primary audio signal and the secondary audio signal in response to a change of the audio error status, and wherein the delay is determined from a characteristic of the music.

2. The audio processor of claim 1 wherein the characteristic of the music comprises the tempo and wherein the secondary audio signal is beat-aligned with the primary audio signal.

3. The audio processor of claim 2 wherein the characteristic of the music comprises the time signature and wherein the secondary audio signal is bar-aligned with the primary audio signal.

4. The audio processor of any preceding claim further comprising a mixer, the mixer comprising a first mixer input coupled to the audio input, a second mixer input, and a mixer output coupled to the audio output, and wherein the mixer is configured to receive a secondary audio signal on the second mixer input and, in response to a change in the error status, to cross-fade between the primary audio signal and the secondary audio signal by modifying the primary audio signal with a primary mixing factor, modifying the secondary audio signal with a secondary mixing factor and combining the modified primary audio signal and the modified secondary audio signal.

5. The audio processor of claim 4 comprising a time delay module having a delay input and a delay output, wherein the delay input is coupled to the audio input and wherein the time delay output is coupled to the second mixer input.

6. The audio processor of claim 4 comprising a time delay module having a delay input and a delay output, wherein the delay input is coupled to the audio output and wherein the time delay module is configured to output the secondary audio signal.

7. The audio processor of claims 5 or 6 further comprising a time delay control module having an output coupled to the time delay module and an input coupled to the audio input, wherein the control module is configured to vary the delay of the time delay module dependent on the characteristic of the music.

8. The audio processor of any of claims 4 to 7 further comprising at least one of a loudness level aligner and spectrum aligner having a respective first input coupled to the primary audio input and a respective second input coupled to the delay module output.

9. The audio processor of claims 4 to 8 wherein the mixer is configured to generate a first mixing factor and a second mixing factor, and the audio mixer is configured to apply the first mixing factor to the primary audio signal and to apply the second mixing factor to the secondary audio signal, and to combine the two signals; wherein in response to the audio error status indicating an error, the first mixing factor changes from a 1 value to 0 value and the second mixing factor changes from a 0 value to a 1 value, and in response to the audio error status indicating no error, the first mixing factor changes from a 0 value to a 1 value and the second mixing factor changes from a 1 value to a 0 value.

10. The audio processor of any preceding claim further comprising an audio decoder configured to receive an audio signal and output the primary audio signal and the audio error status.

11. A method of audio processing for a digital radio, the method comprising

receiving a primary audio signal comprising music;
receiving an audio error status of the primary audio signal;
5 generating a secondary audio signal comprising a delayed version of the primary audio signal, the delay being determined from a characteristic of the music; and
generating an audio output signal by cross-fading between the primary audio signal and the secondary audio signal in response to a change in audio error status.

12. The method of claim 11 wherein generating the secondary audio signal further comprises applying level alignment to substantially align the level of the secondary audio signal with the primary audio signal.

13. The method of any of claims 11 or 12 wherein generating the secondary audio signal further comprises applying spectrum alignment to substantially align the spectrum of the secondary audio signal with the primary audio signal.

14. The method of any of claims 11 to 13 wherein generating the secondary audio signal comprises applying a time delay to the primary audio signal.

15. The method of any of claims 11 to 13 wherein generating the secondary audio signal comprises applying a time delay to the audio output signal.

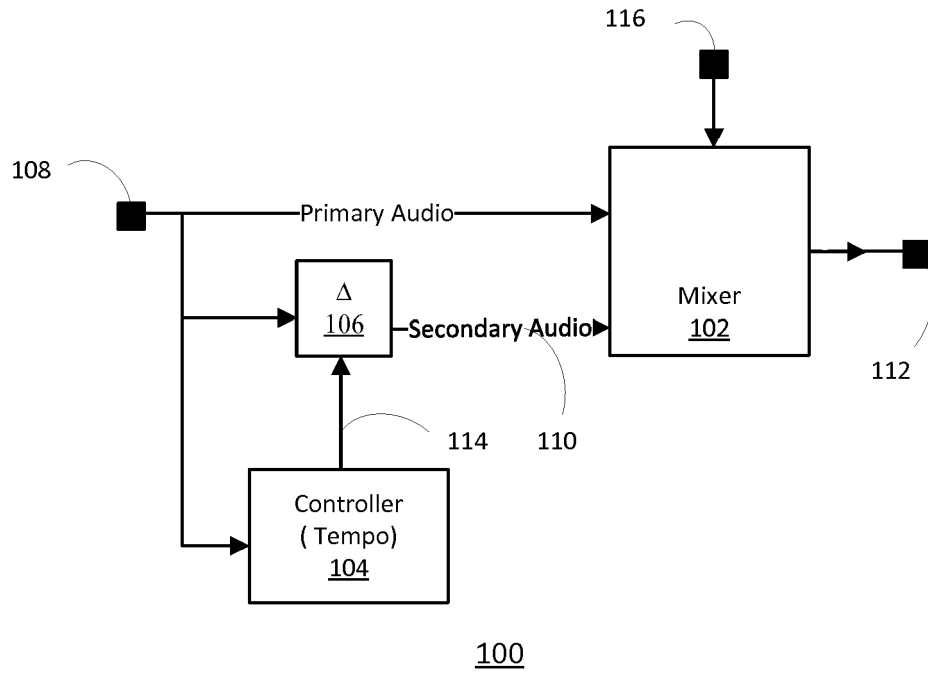


FIGURE 1

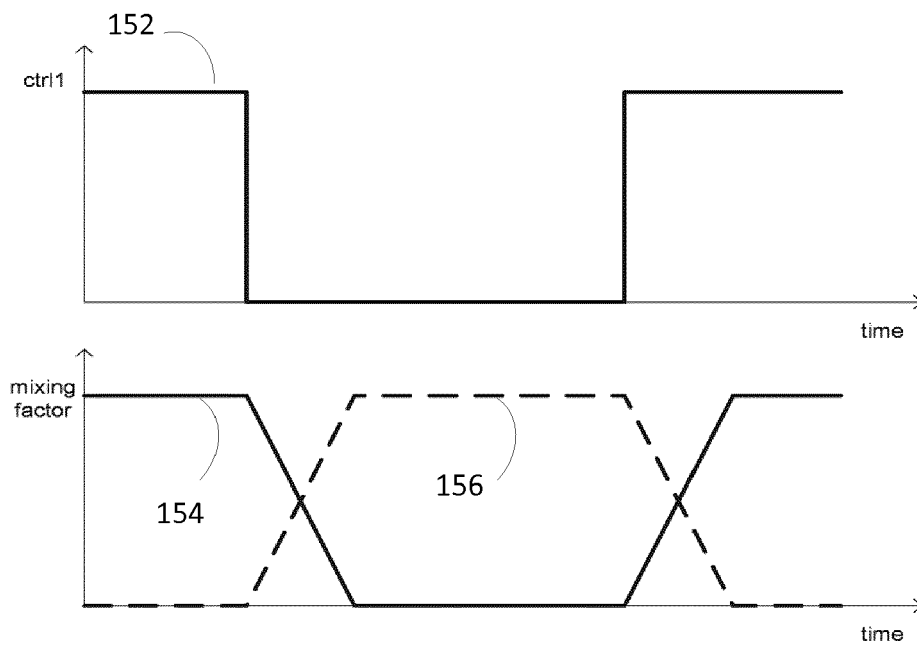


FIGURE 2

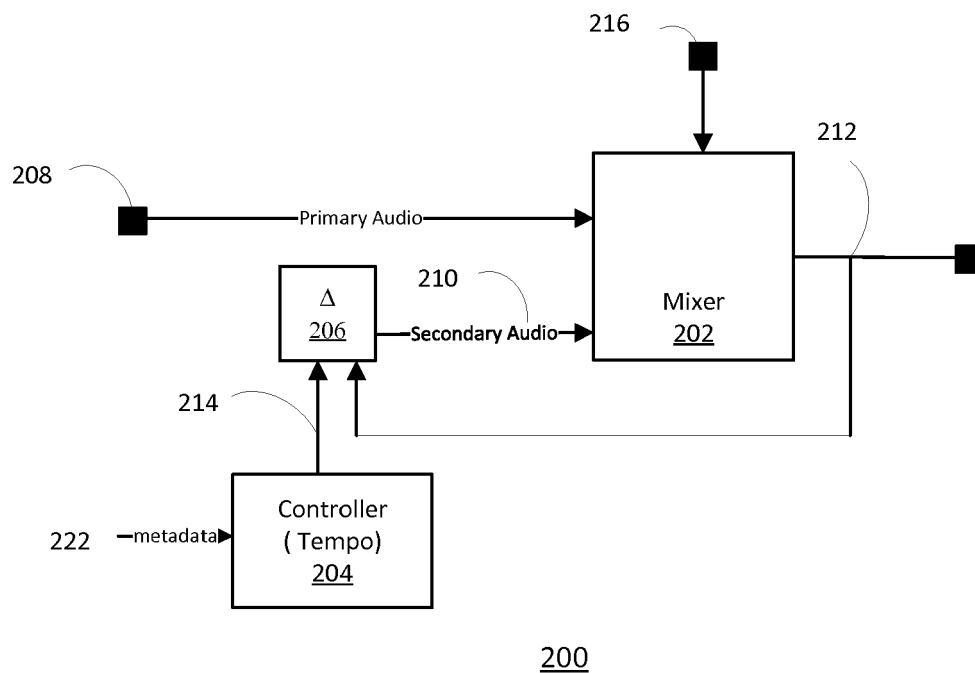


FIGURE 3

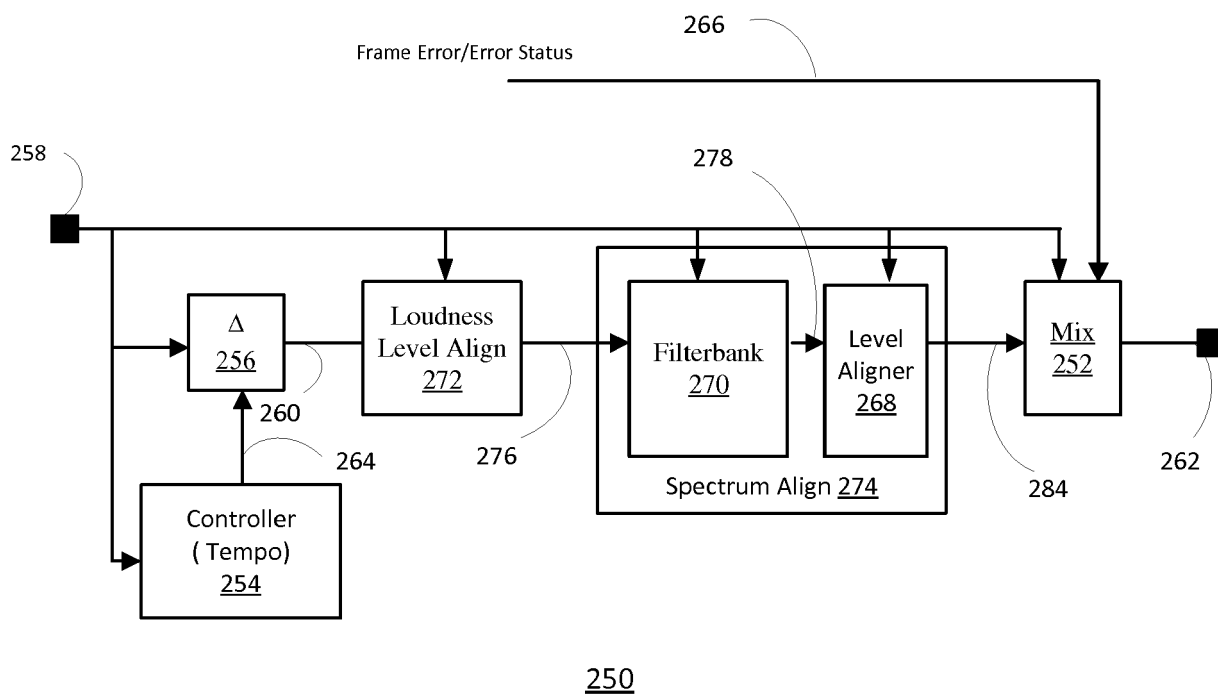
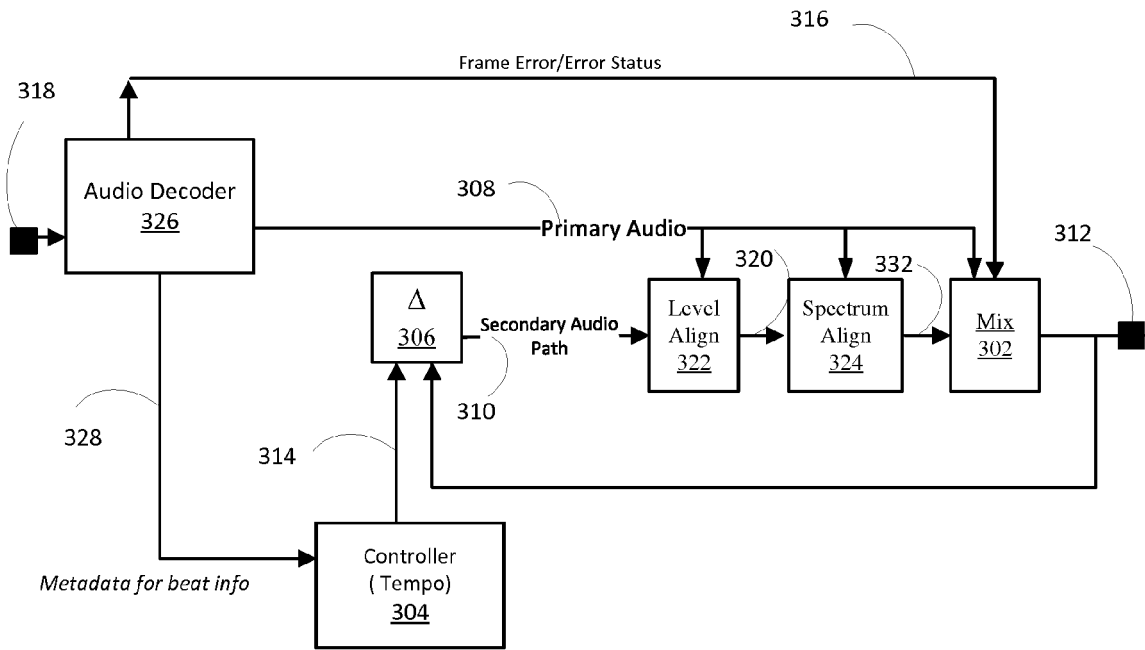
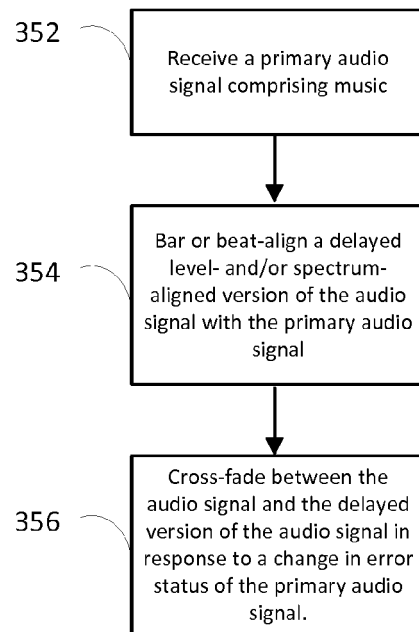


FIGURE 4



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FIGURE 5



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FIGURE 6



EUROPEAN SEARCH REPORT

Application Number
EP 17 16 5312

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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	US 2002/133764 A1 (WANG YE [FI]) 19 September 2002 (2002-09-19) * paragraphs [0016], [0034] - [0037], [0067] - [0070] * * figures 1,4,7 * -----	1-15	INV. H04H60/11
			TECHNICAL FIELDS SEARCHED (IPC)
			H04H
The present search report has been drawn up for all claims			
Place of search The Hague		Date of completion of the search 3 October 2017	Examiner Iovescu, Vladimir
CATEGORY OF CITED DOCUMENTS X : particularly relevant if taken alone Y : particularly relevant if combined with another document of the same category A : technological background O : non-written disclosure P : intermediate document T : theory or principle underlying the invention E : earlier patent document, but published on, or after the filing date D : document cited in the application L : document cited for other reasons & : member of the same patent family, corresponding document			

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