

CLAIMS

1. A method of generating a substitution frame for a lost audio frame of an audio signal, the method comprising
- 5 • determining an audio filter based on samples of a valid audio frame preceding the lost audio frame; and
- generating the substitution frame based on the audio filter and the samples of the valid audio frame preceding the lost audio frame.
- 10 2. The method according to claim 1, wherein the step of generating the substitution frame based on the audio filter and the samples of the valid audio frame includes
- initializing a filter memory of the audio filter with the samples of the valid audio frame.
3. The method according to claim 1 or 2, further comprising:
- 15 • determining a modified audio filter based on the audio filter,
- wherein the modified audio filter replaces the audio filter and wherein the generating of the substitution frame based on the audio filter includes generating the substitution frame based on the modified audio filter and the samples of the valid audio frame.
- 20 4. The method according to claim 3, wherein the step of determining the modified audio filter includes bandwidth sharpening.
5. The method according to any one of the preceding claims, wherein the audio filter is an all-pole filter.
- 25 6. The method according to any one of the preceding claims, wherein the audio filter is derived from an all-pass filter operated on at least a sample of a valid frame.
7. The method according to claim 6, comprising
- 30 • determining the audio filter based on a denominator polynomial of a transfer function of the all-pass filter.
8. The method according to any one of the preceding claims depending on claim 4, wherein the bandwidth sharpening is applied such that a duration of an impulse response of the modified audio filter is extended with regard to a duration of an impulse response of the audio filter.
- 35 9. The method according to any one of the preceding claims depending on claim 4, wherein the bandwidth sharpening is applied such that a distance between a pole of the modified audio filter and the

unit circle is reduced compared to a distance between a corresponding pole of the audio filter and the unit circle.

10. The method according to any one of the preceding claims depending on claim 4, wherein the
5 bandwidth sharpening is applied such that a pole of the modified audio filter with the largest magnitude is equal to 1 or at least close to 1.

11. The method according to any one of the preceding claims depending on claim 4, wherein the
bandwidth sharpening is applied such that a frequency of a pole of the modified audio filter with the
10 largest magnitude is equal to a frequency of a pole of the audio filter with the largest magnitude.

12. The method according to any one of the preceding claims, comprising

- determining the magnitudes and frequencies of the poles of the audio filter using a root-finding method.

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13. The method according to any one of the preceding claims depending on claim 4, wherein the bandwidth sharpening is applied such that the magnitudes of the poles of the modified audio filter are set equal to 1 or at least close to 1, wherein the frequencies of the poles of the modified audio filter are identical to the frequencies of the poles of the audio filter.

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14. The method according to any one of the preceding claims depending on claim 4, wherein a magnitude of a pole of the modified audio filter is set equal to 1 or at least close to 1 only if a magnitude of the corresponding pole of the audio filter has a magnitude exceeding a certain threshold value.

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15. The method according to any one of the preceding claims, wherein the audio filter is a linear predictive coding (LPC) synthesis filter.

16. The method according to any one of the preceding claims depending on claim 3, wherein the method comprises

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- determining filter coefficients of the audio filter;
 - applying the bandwidth sharpening using a bandwidth sharpening factor such that $S_\gamma(z) = S(z/\gamma)$, wherein S_γ denotes a transfer function of the modified audio filter, S denotes a transfer function of the audio filter, and γ denotes the bandwidth sharpening factor; and
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- generating the substitution frame based on the filter coefficients of the audio filter, the samples of the valid audio frame preceding the lost audio frame, and the bandwidth sharpening factor γ .

17. The method according to claim 16, wherein the bandwidth sharpening factor is determined in an iterative procedure by stepwise incrementing and/or decrementing the bandwidth sharpening factor.
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18. The method according to claim 17, further comprising
- checking whether a pole of the modified audio filter lies within the unit circle by converting polynomial coefficients of the modified audio filter to reflection coefficients.
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19. The method according to claim 18, wherein converting the polynomial coefficients of the modified audio filter to reflection coefficients is based on the backward Levinson recursion.
20. The method according to any one of claims 16 to 19, wherein the bandwidth sharpening factor is determined such that a pole of the modified audio filter with the largest magnitude is moved as close to the unit circle as possible, and, at the same time, all poles of the modified audio filter are located within the unit circle.
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21. The method according to any one of claims 16 to 20, wherein the substitution frame is generated using the equation $\hat{x}(n) = \sum_{i=1}^P a_i \cdot \gamma^i \hat{x}(n-i)$, $n \geq 0$, wherein a_i denotes the filter coefficients of the audio filter, P denotes the order of the audio filter, γ denotes the bandwidth sharpening factor, $\hat{x}(-1 \dots -P)$ denotes the filter memory of the audio filter, and $\hat{x}(n)$, $n \geq 0$ denote substitution samples of the substitution frame.
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22. The method according to any one of the preceding claims depending on claim 3, wherein the method comprises
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- determining filter coefficients of the audio filter;
 - applying the bandwidth sharpening by reducing the distance of a pair of line spectral frequencies representing the audio filter coefficients, thereby generating modified line spectral frequencies;
 - 30 • deriving the coefficients of the modified audio filter from the modified line spectral frequencies; and
 - generating the substitution frame based on the filter coefficients of the modified audio filter and the samples of the valid audio frame preceding the lost audio frame.
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23. The method according to any one of the preceding claims, wherein the lost audio packet is associated with a low frequency effect LFE channel of a multi-channel audio signal.

24. The method according to any one of the preceding claims, wherein the lost audio packet has been transmitted over wireless channel from a transmitter to a receiver, and wherein the method is carried out at the receiver.

5 25. The method according to any one of the preceding claims, comprising

- downsampling the samples of the valid audio frame before generating substitution samples of the substitution frame, and
- upsampling the substitution samples of the substitution frame after generating the substitution frame.

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26. The method according to any one of the preceding claims, wherein a plurality of audio frames is lost, comprising

- determining a first modified audio filter by scaling audio filter coefficients of the audio filter using a first bandwidth sharpening factor,
- 15 • determining a second modified audio filter by scaling said audio filter coefficients using a second bandwidth sharpening factor,
- generating substitution frames based on the first modified audio filter for the first M lost audio frames, and
- 20 • generating substitution frames based on the second modified audio filter for the (M+1)th lost audio frame and all following lost audio frames such the audio signal is damped for the latter frames.

27. The method according to any one of the preceding claims, comprising

- splitting the audio signal into a first subband signal and a second subband signal,
- 25 • generating a first subband audio filter for the first subband signal,
- generating first subband substitution frames based on the first subband audio filter,
- generating a second audio filter for the second subband signal,
- generating second subband substitution frames based on the second subband audio filter,
- generating the substitution frame by combining the first and the second subband substitution frames.
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28. The method according to any one of the preceding claims, wherein the audio filter is configured to operate as a resonator,

- the resonator being tuned on the samples of the valid audio frame preceding the lost audio frame;
- 35 • the resonator initially being excited with at least one sample among the samples of the valid audio frame preceding the lost audio frame; and

- the substitution frame is generated by using ringing of the resonator for extending the at least one sample into the lost audio frame.
29. A system comprising:
- 5 one or more processors; and
- a non-transitory computer-readable medium storing instructions that, when executed by the one or more processors, cause the one or more processors to perform operations of any claim of claims 1 to 28.
- 10 30. A non-transitory computer-readable medium storing instructions that, when executed by one or more processors, cause the one or more processors to perform operations of any claim of claims 1 to 28.