CLAIMS

1. An audio processing system (600; 700) comprising a multiband filter (660; 770) for providing a partially complex frequency-domain representation of a signal, the multiband filter comprising:

a synthesis stage (691; 710; 810, 870) receiving a first subband range of a first frequency-domain representation of a signal, the first frequency-domain representation being segmented into time blocks and comprising first spectral components representing spectral content of the signal in the first subband range expressed in a first subspace of a multidimensional space, and outputting, based on the first subband range, an intermediate time-domain representation of the signal;

an analysis stage (693; 720; 820, 880) receiving the intermediate time-domain representation of the signal and outputting, based thereon, a second frequency-domain representation of the signal, the second frequency-domain representation being segmented into time blocks and comprising second spectral components representing spectral content of the signal in the first subband range expressed in a second subspace of the multidimensional space that includes a portion of the multidimensional space not included in the first subspace; and

a processor (640; 740; 860) receiving the first and second subband ranges of the first frequency-domain representation of the signal and the second frequency-domain representation of the signal and combining these to output a partially complex frequency-domain representation of the signal,

wherein:

the synthesis stage is operable to release an approximate value of the intermediate time-domain representation in a time block located $d_1 \geq 1$ time blocks ahead of its output block, which approximate value is computed on the basis of any available time blocks of the first frequency-domain representation; and

said approximate value contributes, in the analysis stage, to a time block of the second frequency-domain representation of the signal.

2. The audio processing system of claim 1, said multiband filter further comprising a transform stage (660; 701) arranged upstream of the multiband filter, said transform stage receiving an input time-domain representation of the signal and outputting the first frequency-domain representation of the signal.

3. The audio processing system of claim 2, wherein the transform stage is one in the group comprising:

a real-valued QMF analysis bank,

a pseudo-QMF analysis bank,

a discrete sine or cosine transform,

DCT-II,

DCT-III,

a modified discrete sine or cosine transform.

4. The audio processing system of any of the preceding claims, wherein the synthesis stage comprises a first finite impulse response filter, FIR, (100; 200) with impulse response $\begin{bmatrix} h_0 & h_1 & h_2 & \dots & h_{L_S} \end{bmatrix}$, where coefficient block $h_0 \neq (0,0,...,0)$.

5. The audio processing system of claim 4, wherein:

the FIR filter comprises one or more output buffers (101, 102, 103, 104, 105) for storing approximate values of the intermediate time-domain representation;

receipt of a new time block of the first frequency-domain representation triggers the FIR filter to increment respective output buffers by the new time block after pre-multiplication by corresponding impulse response coefficient blocks; and

the synthesis stage permits the analysis stage to access the buffer storing the approximate value of the intermediate time-domain representation in a time block located d_1 time blocks ahead of the output block.

6. The audio processing system of claim 5, wherein the approximate value of the intermediate time-domain representation is computed as if any non-available time block of the first frequency-domain representation were zero.

7. The audio processing system of claim 4, wherein the FIR filter comprises:

one or more input buffers (201, 202, 203, 204) for storing recent time blocks of the first frequency-domain representation; and

a weighted summer (221) reading out fewer than $L_S + 1$ input buffers, applying a subset of the impulse response coefficient blocks and outputting an approximate

value of the intermediate time-domain representation in a time block located d_1 time blocks ahead of the output block.

8. The audio processing system of any of claims 5 to 7, wherein the computation of the approximate value includes applying impulse response coefficient blocks representing at least 50 % of the total impulse response mass.

9. The audio processing system of any of claims 5 to 7, wherein the computation of the approximate value includes applying a sequence of consecutive impulse response coefficient blocks $\begin{bmatrix} h_p & h_{p+1} & h_{p+2} & \dots & h_{L_S} \end{bmatrix}$, where $p \geq 1$, which sequence includes the local absolute maximum of the impulse response.

10. The audio processing system of any of the preceding claims, wherein the analysis stage comprises a second finite impulse response filter, FIR, (400; 500) with impulse response $\begin{bmatrix} g_0 & g_1 & g_2 & \dots & g_{L_A} \end{bmatrix}$, where coefficient block $g_0 \neq (0,0,\dots,0)$.

11. The audio processing system of claim 10, wherein:

the second FIR filter comprises one or more output buffers for storing approximate values of the second frequency-domain representation;

receipt, from the synthesis stage, of a new time block of the intermediate time-domain representation triggers the FIR filter to increment a first subset of the output buffers by the new time block after pre-multiplication by corresponding impulse response coefficient blocks; and

receipt, from the synthesis stage, of an approximate value of the intermediate time-domain representation triggers the FIR filter to increment a second subset of the output buffers, which includes the output buffer corresponding to the output block, by the approximate value after pre-multiplication by the corresponding impulse response coefficient block.

12. The audio processing system of claim 10, wherein the second FIR filter comprises:

one or more input buffers (401, 402, 403, 404) for storing recent time blocks of the intermediate time-domain representation;

a weighted summer for reading out fewer than $L_A + 1$ input buffers, applying a subset of the impulse response coefficient blocks, adding the approximate value after pre-multiplication with the corresponding impulse response coefficient block and outputting this as an output block.

13. The audio processing system of any of claims 10 to 12, wherein the computation of the approximate value includes applying impulse response coefficient blocks representing at least 50 % of the total impulse response mass.

14. The audio processing system of any of claims 10 to 12, wherein the computation of the approximate value includes applying a sequence of consecutive impulse response coefficient blocks $\begin{bmatrix} g_p & g_{p+1} & g_{p+2} & \dots & g_{L_S} \end{bmatrix}$, where p ≥ 1, which sequence includes the local absolute maximum of the impulse response.

15. The audio processing system of any of the preceding claims, further comprising a first delay line (630; 730; 830, 890) receiving a second subband range of the first frequency-domain representation of the signal and synchronizing the first frequency-domain representation with the second frequency-domain representation of the signal.

16. The audio processing system of any of the preceding claims, further comprising a second delay line (750) receiving a first subband range of the first frequency-domain representation of the signal and synchronizing the first subband of the first frequency-domain representation with the second frequency-domain representation.

17. The audio processing system of claim 15 or 16, wherein at least one of the delay lines (730, 750) are configured to achieve the synchronization by performing one of the following operations:

a) temporarily storing its received signal;

b) time stamping its received signal;

c) forming a data structure comprising a time block of its received signal and a synchronous time block of the other signal included in the synchronization.

18. The audio processing system of any of the preceding claims, wherein the first subband range is a relatively lower frequency range and the second subband range is a relatively upper frequency range.

19. The audio processing system of any of the preceding claims, wherein the system is an audio encoder.

20. The audio processing system of any of the preceding claims, wherein the system is an audio decoder.

21. An audio processing method for providing a partially complex frequency-domain representation of a signal, comprising the steps:

receiving a first subband range of a first frequency-domain representation of a signal, the first frequency-domain representation being segmented into time blocks and comprising first spectral components representing spectral content of the signal in the first subband range expressed in a first subspace of a multidimensional space;

generating, based on the first subband range, an intermediate time-domain representation of the signal;

generating, based on the intermediate time-domain representation a second frequency-domain representation of the signal, the second frequency-domain representation being segmented into time blocks and comprising second spectral components representing spectral content of the signal in the first subband range expressed in a second subspace of the multidimensional space that includes a portion of the multidimensional space not included in the first subspace;

synchronizing the first frequency-domain representation with the second frequency-domain representation of the signal; and

combining the first and second subband ranges of the first frequency-domain representation of the signal and the second frequency-domain representation of the signal to output a partially complex frequency-domain representation of the signal,

wherein the step of generating the second frequency-domain representation includes using an approximate value of the intermediate time-domain representation in a time block located $d_1 \geq 1$ time blocks ahead of the earliest time block in which a set of time blocks of the first frequency-domain representation, sufficient for an exact computation of the same time block, would have been available, which approximate

value is computed on the basis of any available time blocks of the first frequency-domain representation.

22. A data carrier comprising computer-readable instructions for performing the method of claim 21.

CLAIMS

1. An audio processing system comprising a multiband filter for providing a partially complex frequency-domain representation of a signal, the multiband filter comprising:

a synthesis stage receiving a first subband range of a first frequency-domain representation of a signal, the first frequency-domain representation being segmented into time blocks and comprising first spectral components representing spectral content of the signal in the first subband range expressed in a first subspace of a multidimensional space, and outputting, based on the first subband range, an intermediate time-domain representation of the signal;

an analysis stage receiving the intermediate time-domain representation of the signal and outputting, based thereon, a second frequency-domain representation of the signal, the second frequency-domain representation being segmented into time blocks and comprising second spectral components representing spectral content of the signal in the first subband range expressed in a second subspace of the multidimensional space that includes a portion of the multidimensional space not included in the first subspace; and

a processor receiving the first and second subband ranges of the first frequency-domain representation of the signal and the second frequency-domain representation of the signal and combining these to output a partially complex frequency-domain representation of the signal,

wherein:

the synthesis stage is operable to release an approximate value of the intermediate time-domain representation in a time block located $d_1 \geq$ time blocks ahead of its output block, which approximate value is computed on the basis of any available time blocks of the first frequency-domain representation; and

said approximate value contributes, in the analysis stage, to a time block of the second frequency-domain representation of the signal.

2. The audio processing system of claim 1, said multiband filter further comprising a transform stage arranged upstream of the multiband filter, said transform stage receiving an input time-domain representation of the signal and outputting the first frequency-domain representation of the signal.

3. The audio processing system of claim 2, wherein the transform stage is one in the group comprising:

a real-valued QMF analysis bank,

a pseudo-QMF analysis bank,

a discrete sine or cosine transform,

DCT-II,

DCT-III,

a modified discrete sine or cosine transform.

4. The audio processing system of claim 1, wherein the synthesis stage comprises a first finite impulse response filter, FIR, with impulse response $\left[h_0 \quad h_1 \quad h_2 \quad \ldots \quad h_{L_S}\right]$, where coefficient block $h_0 \neq (0,0,...,0)$.

5. The audio processing system of claim 4, wherein:

the FIR filter comprises one or more output buffers for storing approximate values of the intermediate time-domain representation;

receipt of a new time block of the first frequency-domain representation triggers the FIR filter to increment respective output buffers by the new time block after pre-multiplication by corresponding impulse response coefficient blocks; and

the synthesis stage permits the analysis stage to access the buffer storing the approximate value of the intermediate time-domain representation in a time block located d_1 time blocks ahead of the output block.

6. The audio processing system of claim 5, wherein the approximate value of the intermediate time-domain representation is computed as if any non-available time block of the first frequency-domain representation were zero.

7. The audio processing system of claim 4, wherein the FIR filter comprises:

one or more input buffers for storing recent time blocks of the first frequency-domain representation; and

a weighted summer reading out fewer than $L_S + 1$ input buffers, applying a subset of the impulse response coefficient blocks and outputting an approximate value of the intermediate time-domain representation in a time block located d_1 time blocks ahead of the output block.

8. The audio processing system of claim 5, wherein the computation of the approximate value includes applying impulse response coefficient blocks representing at least 50% of the total impulse response mass.

9. The audio processing system of claim 5, wherein the computation of the approximate value includes applying a sequence of consecutive impulse response coefficient blocks $[h_p \quad h_{p+1} \quad h_{p+2} \quad \dots \quad h_{L_S}]$, where $p \geq 1$, which sequence includes the local absolute maximum of the impulse response.

10. The audio processing system of claim 1, wherein the analysis stage comprises a second finite impulse response filter, FIR, with impulse response $[g_0 \quad g_1 \quad g_2 \quad \dots \quad g_{L_A}]$, where coefficient block $g_0 \neq (0,0,\dots,0)$.

11. The audio processing system of claim 10, wherein:

the second FIR filter comprises one or more output buffers for storing approximate values of the second frequency-domain representation;

receipt, from the synthesis stage, of a new time block of the intermediate time-domain representation triggers the FIR filter to increment a first subset of the output buffers by the new time block after pre-multiplication by corresponding impulse response coefficient blocks; and

receipt, from the synthesis stage, of an approximate value of the intermediate time-domain representation triggers the FIR filter to increment a second subset of the output buffers, which includes the output buffer corresponding to the output block, by the approximate value after pre-multiplication by the corresponding impulse response coefficient block.

12. The audio processing system of claim 10, wherein the second FIR filter comprises:

one or more input buffers for storing recent time blocks of the intermediate time-domain representation;

a weighted summer for reading out fewer than $L_A + 1$ input buffers, applying a subset of the impulse response coefficient blocks, adding the approximate value after pre-multiplication with the corresponding impulse response coefficient block and outputting this as an output block.

13. The audio processing system of claim 10, wherein the computation of the approximate value includes applying impulse response coefficient blocks representing at least 50% of the total impulse response mass.

14. The audio processing system of claim 10, wherein the computation of the

approximate value includes applying a sequence of consecutive impulse response coefficient blocks $[g_p \quad g_{p+1} \quad g_{p+2} \quad \ldots \quad g_{L_S}]$, where p ≥ 1, which sequence includes the local absolute maximum of the impulse response.

15. The audio processing system of claim 1, further comprising a first delay line receiving a second subband range of the first frequency-domain representation of the signal and synchronizing the first frequency-domain representation with the second frequency-domain representation of the signal.

16. The audio processing system of claim 1, further comprising a second delay line receiving a first subband range of the first frequency-domain representation of the signal and synchronizing the first subband of the first frequency-domain representation with the second frequency-domain representation.

17. The audio processing system of claim 15, wherein at least one of the delay lines are configured to achieve the synchronization by performing one of the following operations:

a) temporarily storing its received signal;

b) time stamping its received signal;

c) forming a data structure comprising a time block of its received signal and a synchronous time block of the other signal included in the synchronization.

18. The audio processing system of claim 1, wherein the first subband range is a relatively lower frequency range and the second subband range is a relatively upper frequency range.

19. The audio processing system of claim 1, wherein the system is an audio encoder or an audio decoder.

20. An audio processing method for providing a partially complex frequency-domain representation of a signal, comprising the steps:

receiving a first subband range of a first frequency-domain representation of a signal, the first frequency-domain representation being segmented into time blocks and comprising first spectral components representing spectral content of the signal in the first subband range expressed in a first subspace of a multidimensional space;

generating, based on the first subband range, an intermediate time-domain representation of the signal;

generating, based on the intermediate time-domain representation a second frequency-domain representation of the signal, the second frequency-domain representation being segmented into time blocks and comprising second spectral components representing spectral content of the signal in the first subband range expressed in a second subspace of the multidimensional space that includes a portion of the multidimensional space not included in the first subspace;

synchronizing the first frequency-domain representation with the second frequency-domain representation of the signal; and

combining the first and second subband ranges of the first frequency-domain representation of the signal and the second frequency-domain representation of the signal to output a partially complex frequency-domain representation of the signal,

wherein the step of generating the second frequency-domain representation includes using an approximate value of the intermediate time-domain representation in a time block located $d_1 \geq 1$ time blocks ahead of the earliest time block in which a set of time blocks of the first frequency-domain representation, sufficient for an exact computation of the same time block, would have been available, which approximate value is computed on the basis of any available time blocks of the first frequency-domain representation.

21. A data carrier comprising computer-readable instructions for performing the method of claim 20.