Filter Conversion

Implementing the integrated QMF domain SPAR decoding and processing, for example as shown in **Fig. 2** or **Fig. 3**, requires conversion of the MDFT SPAR filters used for encoding into the QMF domain (e.g., via the filter conversion operator of equation (8), $H = QMF_C\{h\}$), or in general, conversion of the filters of a first filter bank domain into a second, different, filter bank domain, for example by means of FIR filtering along time in the bands of the second filter bank domain.

An example of filter conversion, for example from (MDFT) SPAR FIR filters to QMF-domain SPAR FIR filters is schematically shown in **Fig. 4**. In this example, the SPAR FIR filters 410 are subjected to FIR to QMF-FIR conversion at block 430, to generate QMF-domain SPAR FIR filters 440. Block 430 may take a set of conversion parameters 420 as additional input. These conversion parameters 420 may include, for example, an indication of the maximum number of QMF-domain taps and/or an indication of a minimum relative coefficient magnitude. Based on the conversion parameters 420, the filter conversion at block 430 may comprise, for example, truncation of filters as detailed below.

Broadly speaking, in the filter conversion for each SPAR filter a set of complex-valued FIR filters is derived, one for each QMF band. There may be 60 QMF bands in total, for example. When applied in the QMF domain, this approximates the operation of FIR filtering with one SPAR filter and subsequent QMF analysis. To mimic parameter modification (e.g., prediction) in all SPAR bands and filter bank synthesis, (e.g., 60) complex-valued FIR filters, one for each QMF band, can be derived by summing (e.g., by filter bank synthesis) over the (e.g., 12) parameter-modified complex-valued FIR filters per QMF band.

For the broadband SPAR FIR to QMF domain FIR conversion, first a new prototype filter is derived based on a least squares error objective based on the QMF prototype, the processing stride, the QMF-analysis-synthesis delay, and number of QMF bands. This new prototype typically may have a length of 3 times the processing stride, for example, and is in general asymmetric. Now the QMF domain complex-valued FIR filters can be computed by running a QMF analysis using this new prototype filter with one SPAR FIR filter as input.

In general, the new prototype filter (filter converter prototype) for filter conversion may be derived based on the prototype of the second filter bank.

Accordingly, a given first filter h_b (with appropriate delay) may be approximated by the first and second elementary signals, and (a subset of) the coefficients a_l and b_l may then be used for deriving the adapted first filter H_l^b in second band l.

Apparatus for Implementing Methods According to the Disclosure

Finally, the present disclosure likewise relates to an apparatus (e.g., computer-implemented apparatus) for performing methods and techniques described throughout the present disclosure. **Fig. 19** shows an example of such apparatus 1900. In particular, apparatus 1900 comprises a processor 1910 and a memory 1920 coupled to the processor 1910. The memory 1920 may store instructions for the processor 1910. The processor 1910 may also receive, among others, suitable input data 1930 (e.g., audio input), depending on use cases and/or implementations. The processor 1910 may be adapted to carry out the methods/techniques described throughout the present disclosure (e.g., method 300 of **Fig. 3**) and to generate corresponding output data 1940 (e.g., a reconstructed multichannel audio signal), depending on use cases and/or implementations.

Summary of the Disclosure

In summary, the present disclosure relates to:

- Filter bank processing of a first filter bank (e.g., SPAR filter bank) within the domain of another, second, filter bank (e.g., QMF filter bank), taking advantages of each of the individual filter banks in terms of time and frequency resolution and processing stride
- Efficient and low delay SPAR FIR filter conversion to the QMF domain, specifically with an asymmetric QMF prototype filter
- Optionally, QMF-band dependent QMF FIR length truncation for complexity reduction
- Optionally, QMF domain FIR length truncation based on a threshold relative to the maximum magnitude for the individual filters
- Combining SPAR filter bank filtering and signal manipulation

Further, techniques according to the present disclosure may have the following characteristics and advantages:

- No need to adapt SPAR filters to QMF banding

Claims

1. A method of decoding a representation of a multichannel audio signal, wherein the representation comprises a first channel and metadata relating to a second channel, and wherein the metadata comprises, for each of a plurality of first bands of a first filter bank, a respective prediction parameter (115) for making a prediction for the second channel based on the first channel in that first band, the method comprising:

applying (S310) a second filterbank with a plurality of second bands to the first channel to obtain, for each of the second bands, a banded version of the first channel in that second band, wherein the second filter bank is different from the first filter bank;

for each of the second bands, generating (S320) a respective time-domain filter based on the prediction parameters and first filters of the first filter bank, the first filters corresponding to the first bands; and

generating a prediction for the second channel based on the banded versions of the first channel and the time-domain filters in the second bands.

2. The method according to claim 1, wherein generating the prediction of the second channel comprises, for each of the second bands:

generating (S330) a prediction for the second channel in that second band based on a filtered version of the first channel in that second band, the filtered version of the first channel being obtained by applying the respective time-domain filter in that second band to the banded version of the first channel in that second band.

3. The method according to any one of the preceding claims, wherein the prediction parameters are SPAR parameters and/or, wherein the first filter bank is a SPAR filter bank comprising FIR band filters and uses an MDFT.

4. The method according to any one of the preceding claims, wherein the second filter bank is a QMF filter bank.

5. The method according to any one of the preceding claims, wherein generating the time-domain filter for a given second band comprises:

generating a plurality of adapted first filters based on respective first filters and a prototype filter for filter conversion.

6. The method according to claim 5, wherein for a given second band l the adapted first filter H_l^b of a first filter h_b for a given first band b is optionally calculated as

$$H_l^b = \sum_n h_b(n + kS)q(n) \exp\left(-i\frac{\pi}{L}\left(l + \frac{1}{2}\right)n\right)$$

where q is the prototype filter for filter conversion, S is the stride of the second filterbank, L is the number of second bands, and summation for n is over the support of the prototype filter q for filter conversion; the method optionally further comprising:

generating the prototype filter for filter conversion based on a prototype filter of the second filterbank, wherein the prototype filter is optionally generated by solving a least-squares problem.

7. The method according to claim 5 or 6, wherein generating the prototype filter for filter conversion comprises:

generating an acausal prototype filter p_A based on the prototype filter p of the second filterbank;

generating a cross-correlation p_2 of the acausal prototype filter p_A and the prototype filter p of the second filterbank;

generating a set of matrices $\boldsymbol{V}^{(k)}, k = -K, \dots, K$ for some integer K with dimensions $S \times R$ and with non-zero elements $v_{n,m}$ only for indices n, m with $n - m$ being an integer multiple of S, where R is the length of the prototype filter for filter conversion; and

solving a set of least-square problems for $\boldsymbol{V}^{(k)}\boldsymbol{q}$, where $\boldsymbol{q}$ is a vector of dimensions $R \times 1$ including the filter coefficients of the prototype filter q for filter conversion.

8. The method according to any one of claims 5 to 7, wherein generating the time-domain filter for a given second band further comprises:

taking a weighted sum of the adapted first filters, wherein the adapted first filters are weighted with prediction coefficients for the respective first bands;

wherein the prototype filter for filter conversion is optionally an asymmetric prototype filter; and

wherein a processing stride for each tap is optionally either equal or smaller than the number of second bands.

9. The method according to any one of claims 1 to 4, wherein generating the time-domain filter for a given second band comprises:

approximating a given first filter by first and second elementary signals,

wherein the first elementary signals are obtainable as results of applying the second filter bank, elementary real-valued single-tap filters, and a synthesis filter bank of the second filter bank to elementary signals with single non-zero samples at respective sample positions, wherein the elementary real-valued single-tap filters are filters for respective single ones of the second bands with single non-zero filter coefficients at respective tap positions; and

wherein the second elementary signals are obtainable as results of applying the second filter bank, elementary imaginary single-tap filters, and the synthesis filter bank of the second filter bank to the elementary signals, wherein the elementary imaginary single-tap filters are filters for respective single ones of the second bands with single non-zero filter coefficients at respective tap positions; and

generating adapted time domain filters for the first filters in the second band based on coefficients of the first and second elementary signals in the approximation.

10. The method according to any one of claims 1 to 4, wherein generating the time-domain filter for a given second band comprises:

obtaining results $u_{p,l,k}$ of applying the second filterbank, real-valued single tap filters $F_{\lambda}^{(re)}(\kappa) = \delta(\lambda - l, \kappa - k)$, and a synthesis filterbank of the second filterbank to signals $x_p(k) = \delta(k - p)$, where l indicates a given second band, p indicates a given sample position, and k indicates a filter tap position;

obtaining results $v_{p,l,k}$ of applying the second filterbank, imaginary single tap filters $F_{\lambda}^{(im)}(\kappa) = i\delta(\lambda - l, \kappa - k)$, and the synthesis filterbank of the second filterbank to the signals $x_p(k) = \delta(k - p)$;

determining a least-squares solution for coefficients a_l and b_l such that

$$\sum_{l=0}^{L-1} \sum_{k=0}^{N_l} \left(a_l(k) u_{p,l,k}(n) + b_l(k) v_{p,l,k}(n) \right) \approx h_b(n - D_3 - p)$$

for a given delay D_3, where h_b is the first filter for first band b, L is the number of second bands, and N_l is a predefined number of filter taps for second band l; and

generating an adapted first filter H_l^b of the first filter h_b in the second band l as $H_l^b = a_l + ib_l$.

11. The method according to any one of the preceding claims, further comprising truncating a filter length of the time-domain filters; wherein the filter length of a given time-domain filter after truncation optionally depends on the respective second band of the time domain filter.

12. The method according to claim 11,

wherein generating the time-domain filter for a given second band involves generating a respective adapted time-domain filter in the given second band for each of the first filters, and generating the time-domain filter in the given second band based on the adapted time-domain filters in the given second band and the prediction parameters; and

wherein truncation of a time-domain filter for the given second band is based on threshold values for the filter coefficients of the adapted time-domain filters, with each threshold value corresponding to a respective one among the first filters, wherein the threshold value for the adapted time-domain filters for a given first filter is derived from a maximum magnitude of said adapted time-domain filters in the plurality of second bands; the method optionally comprising:

determining, for each first band, a maximum magnitude of the corresponding adapted time-domain filters in the plurality of second bands;

for each first band, determining a minimum truncated filter length for the corresponding adapted time-domain filters in the plurality of second bands based on a threshold value derived from said maximum magnitude; and

for each second band, determining the filter length of the time-domain filter in that second band based on the minimum truncated filter lengths of the adapted time-domain filters in that second band.

13. The method according to claim 11, wherein generating the time-domain filter for a given second band comprises:

determining (S1510, S1610) a first band among the plurality of first bands that has a highest energy in that second band; and

generating (S1520, S1620) the time-domain filter based on one of:

a linear-phase approximation of the first filter corresponding to the determined first band and a corresponding prediction coefficient for the determined first band; and

a weighted sum of linear-phase approximations of the first filters corresponding to the determined set of first bands, wherein weights in the weighted sum depend on the corresponding prediction coefficients for the determined set of first bands and respective normalized magnitudes or energies of the first bands of the determined set of first bands in that second band.

14. An apparatus, comprising a processor and a memory coupled to the processor, and storing instructions for the processor, wherein the processor is adapted to carry out the method according to any one of claims 1 to 13.

15. A computer-readable storage medium storing a program comprising instructions that, when executed by a processor, cause the processor to carry out the method according to any one of claims 1 to 13.