CLAIMS

1. A method for processing an audio signal ~~(100)~~, the method comprising:

using linear predictive filtering for removing ~~(S102, S104, S300-S308, S400-S402)~~ a discontinuity ~~(106a, 106b)~~ between a filtered past frame and a filtered current frame of the audio signal,

wherein the method comprises filtering the current frame of the audio signal and removing the discontinuity by modifying a beginning portion of the filtered current frame by a signal obtained by linear predictive filtering a predefined signal with initial states of the linear predictive filter defined on the basis of a last part of the unfiltered past frame filtered using the set of filter parameters for filtering the current frame.

2. The method of claim 1, further comprising estimating the linear predictive filter on the filtered or non-filtered audio signal ~~(100)~~.

3. The method of claim 2, wherein estimating the linear predictive filter comprises estimating the filter based on the past and/or current frame of the audio signal ~~(100)~~ or based on the past filtered frame of the audio signal ~~(100)~~ using the Levinson-Durbin algorithm.

4. The method claim 1, wherein the linear predictive filter comprises a linear predictive filter of an audio codec.

5. The method of one of claims 1 to 4, wherein removing the discontinuity comprises processing the beginning portion of the filtered current frame, wherein the beginning portion of the current frame has a predefined number of samples being less or equal than the total number of samples in the current frame, and wherein processing the beginning portion of the current frame comprises subtracting ~~(S304, S402)~~ a beginning portion of a zero-input-response (ZIR) from the beginning portion of the filtered current frame.

6. The method of claim 5, comprising filtering ~~(S400)~~ the current frame of the audio signal using a non-recursive filter, like a FIR filter, for producing the filtered current frame.

7. The method of claim 5, comprising processing the unfiltered current frame of the

audio signal on a sample-by-sample basis using a recursive filter, like an IIR filter, and wherein processing a sample of the beginning portion of the current frame comprises:

filtering ~~(S302)~~ the sample with the recursive filter using the filter parameters of the current frame for producing a filtered sample, and

subtracting ~~(S304)~~ a corresponding ZIR sample from the filtered sample for producing the corresponding sample of the filtered current frame.

8. The method of claim 7, wherein filtering ~~(S302)~~ and subtracting ~~(S304)~~ are repeated until the last sample in the beginning portion of the current frame is processed, and wherein the method further comprises filtering ~~(S306)~~ the remaining samples in the current frame with the recursive filter using the filter parameters of the current frame.

9. The method of one of claims 5 to 8, comprising generating the ZIR, wherein generating the ZIR comprises:

filtering ~~(S200)~~ the M last samples of the unfiltered past frame with the filter and the filter parameters used for filtering the current frame for producing a first portion of filtered signal, wherein M is the order of the linear predictive filter,

subtracting ~~(S202)~~ from the first portion of filtered signal the M last samples of the filtered past frame, filtered using the filter parameters of the past frame, for generating a second portion of filtered signal, and

generating ~~(S204)~~ a ZIR of a linear predictive filter by filtering a frame of zero samples with the linear predictive filter and initial states equal to the second portion of filtered signal.

10. The method of claim 9, comprising windowing the ZIR such that its amplitude decreases faster to zero.

11. A non-transitory computer program product comprising a computer readable medium storing instructions which, when executed on a computer, carry out the method of one of claims 1 to 10.

12. An apparatus for processing an audio signal ~~(100)~~, the apparatus comprising:

a processor ~~(102, 110, 112)~~ configured to use linear predictive filtering for removing a discontinuity between a filtered past frame and a filtered current frame of the audio signal ,

wherein the processor is configured to filter the current frame of the audio signal and remove the discontinuity by modifying a beginning portion of the filtered current frame by a signal obtained by linear predictive filtering a predefined signal with initial states of the linear predictive filter defined on the basis of a last part of the unfiltered past frame filtered using the set of filter parameters for filtering the current frame.

13. An audio decoder ~~(250)~~, comprising an apparatus of claim 12.

14. An audio encoder ~~(200)~~, comprising an apparatus of claim 12.