Claims

1. A system for filtering an information input signal, divided into different update intervals, according to parameters varying with the update intervals, to obtain a filtered output signal, the system comprising:

a first filter unit to filter a first filter input signal at least at an initial subinterval in a current update interval to obtain a first filter output signal, according to parameters associated to the preceding update interval, the first filter unit being configured to scale the parameters along at least the initial subinterval , wherein the first filter unit is configured to scale the parameters associated to the preceding update interval by a first scaling factor changing towards 0, or a value close to 0, along at least the initial subinterval; and

a second filter unit to filter a second filter input signal, at the initial subinterval, according to parameters associated to the current update interval to obtain a second filter output signal, the second filter unit being configured to scale the parameters along at least the initial subinterval, wherein the second filter unit is configured to scale the parameters associated to the current update interval by a second scaling factor changing from 0, or a value close to 0, toward a value different from 0, or a value more distant from 0 than the value close to 0, along at least the initial subinterval,

wherein the first filter input signal is based on the information input signal, the first filter output signal is an intermediate signal, the second filter input signal is based on the intermediate signal, and the filtered output signal is based on the second filter output signal.

2. The system of claim 1, further comprising:

a third filter unit to filter the information input signal, at a subsequent subinterval in the current update interval after the initial subinterval according to the parameters associated to the current update interval.

3. The system of claim 2, further comprising:

a selector configured to check if the distance between parameters for the current update interval is 0 or within a first threshold, so as to filter the information input signal at least in the initial subinterval using the third filter unit.

4. The system of claim 2 or 3, wherein the third filter unit is configured, along the subsequent interval, to maintain the filter parameters constant.

5. The system of claim 2 or 3 or 4, wherein the third filter unit operates as long term prediction, LTP, pre-filter, LTP post-filter, harmonic LTP pre-filter, or harmonic LTP post-filter.

6. The system of any of claims 2 to 5, further configured to check the gain g_k of the current k^{th} update interval and the gain g_{k-1} of the previous $(k-1)^{th}$ update interval, so that:

- If $g_{k-1}=0$ and $g_k=0$, then the filtering at the first, second, and third filter units is deactivated; and/or
- If $g_{k-1}=0$ and $g_k\neq0$, then
    - The filtering at the first filter unit is deactivated;
    - there is filtering at the second filter unit in at least the initial subinterval;
    - there is filtering at the third filter unit in the subsequent subinterval; and/or
- If $g_{k-1}\neq0$ and $g_k=0$, then
    - there is filtering at the first filter unit in at least the initial subinterval;
    - the filtering at the second filter unit is deactivated;
    - the filtering at the third filter unit is deactivated; and/or
- If $g_{k-1}\neq0$ and $g_k\neq0$, then the difference of the integer and fractional parts of the pitch lag are checked, so that:
    - If the integer and fractional parts of the pitch lag in the current k^{th} update interval and in the previous $(k-1)^{th}$ update interval are the same, then:
        - the filtering at the first filter unit and the second filter units is deactivated;
        - there is filtering at the third filter unit along the whole current update interval;
    - else if there is a difference in the integer or in the fractional part of the pitch lag:
        - there is filtering at the first filter unit in at least the initial subinterval;
        - there is filtering at the second filter unit in at least the initial subinterval
        - there is filtering at the third filter unit in the subsequent subinterval.

7. The system of any of the preceding claims, wherein at least one of the first and second filter units operates as long term prediction, LTP, pre-filter, LTP post-filter, harmonic LTP pre-filter, or harmonic LTP post-filter.

8. The system of any of the preceding claims, wherein at least one of the first and second filter unit has a transfer function comprising a numerator and a denominator, wherein the numerator comprises a gain value indicated by the gain information, and wherein the denominator comprises an integer part of a pitch lag indicated by the pitch lag information and a multi-tap filter depending on a fractional part of the pitch lag.

9. The system of any of the preceding claims, wherein:

the parameters of at least one of the first and second filter unit are obtained from harmonicity information, gain information, pitch lag information, the integer part of the pitch lag of the information input signal and/or the fractional part of the pitch lag of the information input signal.

10. The system of any of the preceding claims, wherein:

the parameters of the first and/or second filter unit are chosen among at least one or a combination of, linear predictive coding, LPC, filter, infinite impulse response, IIR, filter, and/or finite impulse response, FIR, filter.

11. The system of any of the preceding claims, wherein:

the first scaling factor and the second scaling factor are non-negative values complementary with each other to a value greater than 0.

12. The system of any of the preceding claims, wherein:

the first scaling factor changes towards 0 towards the final extremity of at least the initial subinterval, and/or

the second scaling factor changes from 0, or a value close to 0, from the initial extremity of the current update interval towards a non-zero value, or a value which is more distant from 0 than the value close to 0.

13. The system of any of the preceding claims, further comprising:

a fourth filter unit configured to filter the information input signal, at least at the initial subinterval, using parameters obtained by interpolating the parameters associated to the current update interval and the parameters associated to the previous update interval.

14. The system of claim 13, wherein the fourth filter unit operates as long term prediction, LTP, pre-filter, LTP post-filter, harmonic LTP pre-filter, or harmonic LTP post-filter.

15. The system of claim 13 or 14, further comprising:

a selector configured to check if the distance between parameters for the current update interval is within a second threshold, so as to filter the information input signal at least in the initial subinterval using the fourth filter unit.

16. The system of claim 15, further configured to actively set the second threshold on the basis of values associated to the signal.

17. The system of claim 16, further configured to:

set the second threshold as a pitch lag distance threshold equal to the minimum between the integer part of the pitch lag at the current update interval and the integer part of the pitch lag at the previous update interval,

so as to use the fourth filter unit when the distance between the integer part of a pitch lag at the current update interval and the integer part of the pitch lag at the previous update interval is less than the pitch lag distance threshold; and/or

so as to use the first and second filter units when the distance between the integer part of the pitch lag at the current update interval and the integer part of the pitch lag at the previous update interval is greater than the pitch lag distance threshold.

18. The system of claim 16 or 17, further configured to:

use a condition associated to the gains of the signal at the current update interval and at the preceding interval,

so as to use the fourth filter unit when both the gains of the signal at the current update interval and at the preceding interval are different from zero, and/or

so as to use the first and second filter units when at least one of the gains of the signal at the current update interval and at the preceding interval is zero.

19. The system of any of the preceding claims, wherein the information input signal is an audio signal.

20. The system of any of the preceding claims, wherein the first filter unit is to provide the first filter output signal as:

$$y'[n] = x[n] + s_{k-1}[n]\left(\sum_{i=0}^{P} b_{k-1,i}x[n-i] - \sum_{j=1}^{Q} a_{k-1,j}y'[n-j]\right), kT \le n < kT + T_l$$

where the scaling factor $s_{k-1}[n]$ changes towards a value close to 0 when n increases, and the second filter unit is to provide the second filter output signal as:

$$y[n] = y'[n] + s_k[n]\left(\sum_{i=0}^{P} b_{k,i}y'[n-i] - \sum_{j=1}^{Q} a_{k,j}y[n-j]\right), kT \le n < kT + T_l$$

where the scaling factor $s_k[n]$ changes from a value close to 0 towards a non-zero value when n increases, where T is the current k[th] update interval, T_l the initial subinterval, n an instant, x[n] the information input signal, $b_{k-1,i}$ and $a_{k-1,j}$ are parameters associated to the previous (k-1)[th] update interval, $a_{k,j}$ and $b_{k,i}$ are parameters associated to the current k[th] update interval, and P and Q are associated to the type of the filter.

21. The system of any of the preceding claims, wherein the first filter unit is configured to provide the first filter output signal in the form of

$$\widehat{x_{ltpf}}'(n) = \hat{x}(n) - \left(1 - \frac{n}{\frac{N_F}{4}}\right)\left[\sum_{k=0}^{L_{num}} c_{num}^{mem}(k)\hat{x}(n-k) + \sum_{k=0}^{L_{den}} c_{den}^{mem}(k, p_{fr}^{mem})\widehat{x_{ltpf}}'\left(n - p_{int}^{mem} + \frac{L_{den}}{2} - k\right)\right]$$
$$for\ n = 0..\frac{N_F}{4}$$

and the second filter unit is configured to provide the filtered output signal in the form of

$$\widehat{x_{ltpf}}(n) = \widehat{x_{ltpf}}'(n) - \frac{n}{\frac{N_F}{4}}\left[\sum_{k=0}^{L_{num}} c_{num}(k)\widehat{x_{ltpf}}'(n-k) + \sum_{k=0}^{L_{den}} c_{den}(k, p_{fr})\widehat{x_{ltpf}}\left(n - p_{int} + \frac{L_{den}}{2} - k\right)\right]$$
$$for\ n = 0..\frac{N_F}{4}$$

wherein $\frac{N_F}{4}$ is the length of the initial subinterval, $\hat{x}(n)$ is the information input signal, $\widehat{x_{ltpf}}'$ is the intermediate signal, $\widehat{x_{ltpf}}(n)$ is the filtered output signal, n is an instant, p_{int}^{mem} and p_{fr}^{mem} are respectively based on the integer part and fractional part of a pitch lag associated to the preceding update interval, p_{int} and p_{fr} are respectively based on the integer part and fractional part of the pitch lag associated to the current update interval, $c_{num}(k)$ is a coefficient based on the gain value for the current update interval, $c_{den}(k, p_{fr})$ is a coefficient based on the gain value for the determined update interval and on the fractional part of the pitch, $c_{num}^{mem}(k)$ is a coefficient based on the gain value for

preceding update interval, $c_{den}^{mem}(k, p_{fr}^{mem})$ is a coefficient based on the gain value for preceding update interval and on the fractional part of the pitch, L_{den} and L_{num} are fixed and/or based on the sampling rate of the information input signal.

22. The system of any of the preceding claims, wherein the time length of the initial subinterval is between the 5% and the 40% of the time length of the current update interval.

23. The system of any of the preceding claims, comprising an encoder side and a decoder side , wherein at least one of the first and second filter units is at the decoder side.

24. The system of any of the preceding claims, comprising an encoder side and a decoder side, wherein at least one of the first and second filter units is at the encoder side.

25. The system of any of the preceding claims, wherein the encoder side comprises:

a parameter estimator configured to estimate the parameters for at least one of the first and second filter unit.

26. The system of any of the preceding claims, further comprising a converter for converting a first representation of the information signal into a second representation of the information signal.

27. The system of any of the preceding claims, wherein the at least a subinterval is the update interval.

28. The system of any of the preceding claims, further configured to:

determine if the first and/or second filter unit will operate as an identity filter; and

in case of determination, bypass the first and/or second filter.

29. The system of any of the preceding claims, wherein the first filter unit is configured to scale the parameters gradually by fading and/or monotonically or strictly monotonically the parameters along at least the initial subinterval.

30. The system of any of the preceding claims, wherein the second filter unit is configured to scale the parameters gradually by fading and/or monotonically or strictly monotonically the parameters along at least the initial subinterval.

31. A method for filtering an information input signal divided into different update intervals, according to parameters associated with the update intervals, to obtain a filtered output signal, the method comprising:

performing a first filtering at least at an initial subinterval of a current update interval according to parameters associated to the preceding update intervals, wherein the parameters along at least the initial subinterval are scaled by a first scaling factor changing towards 0, or a value close to 0, along at least the initial subinterval; and

performing a second filtering at least at the initial subinterval, according to parameters associated to the current update interval, wherein the parameters along the initial subinterval are scaled by a second scaling factor changing from 0, or a value close to 0, toward a value different from 0, or a value more distant from 0 than the value close to 0, along at least the initial subinterval,

wherein the first filtering is performed on the information input signal and the second filtering is performed on the signal obtained by the first filtering.

32. A non-transitory storage unit storing instructions which, when executed by a processor, cause the processor to perform the method of claim 31.