REAL-TIME IMPLEMENTATION OF A LINEAR-PHASE OCTAVE GRAPHIC EQUALIZER

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ABSTRACT

This paper proposes a real-time implementation of a linear-phase octave graphic equalizer (GEQ), previously introduced by the same authors. The structure of the GEQ is based on interpolated finite impulse response (IFIR) filters and is derived from a single prototype FIR filter. The low computational cost and small latency make the presented GEQ suitable for real-time applications. In this work, the GEQ has been implemented as a plugin of a specific software, used for real-time tests. The performance of the equalizer has been evaluated through subjective tests, comparing it with a filterbank equalizer. For the tests, four standard equalization curves have been chosen. The experimental results show promising outcomes. The result is an accurate real-time-capable linear-phase GEQ with a reasonable latency.

1. INTRODUCTION

In audio applications, graphic equalizers (GEQs) play an essential role. This type of equalizer is called graphic because the user can adjust only the gain of each band, defining a graph of the desired magnitude response [1, 2, 3]. GEQs are realized as filterbanks in which the center frequency and the frequency bandwidth are fixed. This paper presents a real-time implementation of the linear-phase octave GEQ proposed in [4].

GEQs can be categorized depending on their phase response as minimum phase or linear phase. Minimum-phase GEQs show the lowest latency and their impulse response is zero before the main peak, avoiding pre-ringing effects. These properties make minimum-phase GEQs suitable for live music applications, where the latency must be as small as possible. However, the minimum phase may affect the audio perception in applications such as multichannel equalization, parallel processing, phase compatibility of audio equipment, and crossover network design, where the linear phase is more suitable.

A linear-phase system preserves the phase of the input signal [5] and prevents phase distortions. As an example, in multichannel audio equalization, the target magnitude response of each channel changes. A non-linear-phase system also produces variations in the phase response, affecting the spatial impression [6]. In addition, phase distortions may be more audible, especially in speech equalization [7].

Minimum-phase GEQs are usually realized with infinite impulse response (IIR) filters arranged in a cascade [8, 1, 9, 10] or a parallel [1, 11, 12, 13] structure, in which each filter has a fixed center frequency and bandwidth and only the gain is variable [14]. In [15, 10, 13], GEQs formed by second-order IIR sections are proposed, reaching a good accuracy with a reduced computational cost in terms of number of operations per sample. In contrast, linear-phase equalizers can be designed with finite impulse response (FIR) filters, which guarantee a linear phase response [3]. Moreover, FIR filters are not affected by numerical problems that may occur with IIR solutions [16]. The first digital FIR GEQs were developed in the 1980s [17, 18, 19] and, as for IIR GEQs, they are usually designed with parallel structures [17, 18, 20].

A linear-phase GEQ can be realized with a single high-order FIR filter that approximates a target frequency response defined by the user, as suggested by McGrath [20]. The target curve can be obtained through the interpolation of the command-gain points [19, 21], since only the gains of the center frequency bands are well defined. The length of the FIR filter corresponds to the length of the lowest band filter, since it is the narrowest and thus has the longest ringing. Usually, a good accuracy at the low frequencies is achieved with a filter length of at least several thousand samples [19, 22, 23, 24]. In addition, the single FIR filter must be completely redesigned every time the gains at the command points change, requiring additional computing, which is unsuited for real-time modifications of the target response.

Different solutions can be found to reduce the computational complexity of FIR GEQs. Frequency-warped FIR filters [25, 26] allow for the reduction of the filter lengths at low frequencies. However, warped FIR filters are obtained through a frequency transformation that uses IIR filters and results in a non-linear phase response. Fast convolution [27] is another approach applied for reducing the computational cost of FIR GEQs [28, 29, 30, 31]. In this case, the equalized signal is obtained by executing complex multiplication of the discrete Fourier transform (DFT) of the input signal, elaborated in frames, with the DFT of the filter's impulse response and anti-transforming of the result. The transformations are performed by implementing the fast Fourier transform (FFT) to save computational cost. Although the FFT-based approach guarantees a linear phase response, the frame-based filtering introduces a high latency [29, 4].

Linear-phase FIR GEQs also can be realized using multirate processing [18, 19, 32, 23]. In this case, the sample rate can be different for each band. In particular, the higher frequency bands use a large sample rate, while the lower frequency bands work at a slower rate. After the filtering, the band signals are upsampled to the original sample rate and summed to obtain the output of the equalizer. The filterbank equalizer (FBEq) of [32], based on multirate filtering, is used in this paper as a comparison for the experiments.

An interesting option for saving computational complexity has been proposed by Hergum [33]: an FIR GEQ is designed using interpolated FIR (IFIR) filters [34]. The use of IFIR filters for the GEQ design has recently been extended in [35, 4]. An IFIR filter is composed of the cascade of two FIR filters, in which the first one is interpolated (stretched) and the second one attenuates the unwanted spectral images produced by the interpolation [34]. IFIR filters have a linear phase and small ripple requiring a computational cost lower than conventional FIR filters. In [33, 35], the IFIR approach is used to develop uniform GEQs, in which all the bands have the same bandwidth. However, standard graphic equalizers use a logarithmic band division [3], which is necessary for audio applications due to the human perception of sound and the nature of music.

In this context, in [4], an efficient linear-phase octave GEQ based on IFIR filters is proposed. The band filters of the GEQ are arranged in a parallel tree structure derived from a prototype FIR filter. The design uses complementary filters to guarantee a flat magnitude response when all the gains are the same. The IFIR GEQ of [4] presents an approximation error of less than ± 1 dB at the command frequencies, similar to the traditional IIR GEQs. The computational cost is much lower than the linear-phase GEQs and slightly higher than the state-of-the-art IIR GEQs. The reduced latency and the improved computational efficiency of this equalizer make it suitable for real-time applications.

In this paper, a real-time implementation of the GEQ of [4] is presented. The low latency and the reduced computational cost make the GEQ suitable for real-time applications. The implementation is realized by using a specific software and the GEQ is developed as a plugin. The performance of the equalizer has been evaluated through subjective listening tests implementing four standard equalization curves and comparing it with the FBEq of [32].

The paper is organized as follows. Section 2 describes the design of the implemented GEQ. Section 3 explains the real-time implementation. Section 4 discusses the experimental results. Finally, Section 5 concludes the paper.

2. GRAPHIC EQUALIZER DESIGN

The linear-phase GEQ proposed by Bruschi et al. [4] is an octave equalizer with the following ten band center frequencies: 31.25 Hz, 62.5 Hz, 125 Hz, 250 Hz, 500 Hz, 1.0 kHz, 2.0 kHz, 4.0 kHz, 8.0 kHz, and 16.0 kHz. The bands are numbered from lowest to highest using index $m = 1, 2, 3, ..., 10$.

The scheme of the implemented linear-phase octave-band GEQ is shown in Fig. 1. This design uses the sample rate of $f_s = 48$ kHz. The GEQ design is based on a tree structure derived from a half-band lowpass prototype FIR filter $H_{\mathrm{LP}}(z)$ with a cutoff frequency of 12 kHz and order $N = 18$, corresponding to a filter length of $N + 1 = 19$ [4]. The filter $H_{\mathrm{LP}}(z)$ is designed using the Kaiser window with $\beta = 4$ [36]. The last band of the equalizer $H_{10}(z)$ is obtained by the complementary highpass filter of the prototype filter as

$$H_{10}(z) = z^{-D} - H_{\mathrm{LP}}(z), \tag{1}$$

where $D = N/2$ is the delay. According to Eq. (1), the filter of the highest band $H_{10}(z)$ is implemented using a delay line and a subtraction, as shown in the top right of Fig. 1. The other bands are obtained with stretched versions of the prototype filter, such as $H_{\mathrm{LP}}(z^2)$ and $H_{\mathrm{LP}}(z^4)$, which are prepared by inserting one or three zero samples between each two coefficients of the prototype FIR filter, respectively [37].

As depicted in Fig. 2, the output signal $Y_m(z)$ of the mth band is obtained from the input signal $X(z)$ as

$$Y_m(z) = H_m(z)X(z), \tag{2}$$

where $H_m(z)$ is the transfer function of the mth band, computed as

$$H_m(z) = z^{-\Delta_m}[z^{-DL_m} - H_{\mathrm{LP}}(z^{L_m})]G_m(z), \tag{3}$$

where the mth interpolation factor L_m is computed as

$$L_m = 2^{(M-m)} = 2^{(10-m)}, \tag{4}$$

and the transfer function $G_m(z)$, which is shown in detail in Fig. 3, is composed of the cascade of all previous band filters as

$$G_m(z) = H_{\mathrm{LP}}(z) \prod_{k=m+1}^{M-1} H_{\mathrm{LP}}(z^{L_k}), \tag{5}$$

with $m = 2, 3, ..., M$ and $M = 10$. Looking at Fig. 2, the input signal $x(n)$ is first filtered by the filter $G_m(z)$ and the resulting intermediate signal $x_m(n)$, shown for each band in Fig. 1, is then filtered by $H_{10}(z^{L_m})$ that is implemented through a delay line and a subtraction, according to Eq. (1).

From the third to the tenth band, a synchronization delay Δ_m, also shown in Fig. 2, is applied to align the band outputs and it is computed as

$$\Delta_m = \tau - [2^{(M+1-m)} - 1]D = \tau - [2^{(11-m)} - 1]D, \tag{6}$$

where τ is the total delay of the equalizer in samples:

$$\tau = [2^{(M-1)} - 1]D = 511D. \tag{7}$$

In Fig. 1, the synchronization delays Δ_m are shown one upon the other on the right-hand side, next to the command gain factors g_m. In the highest band (the top signal path in Fig. 1), the total delay of $511D$ samples is formed by the cascade of the delay line z^{-D} and the synchronization delay z^{-510D}. In the lowest band, the synchronization delay is formed by the cascade of all the delay lines between the input (top left corner in Fig. 1) and the output y_1 (bottom right corner in Fig. 1), which have the lengths D, $2D$, $4D$, $8D$, $16D$, $32D$, $64D$, $128D$, and $256D$. This adds up to $511D$ samples of delay.

The lowest band filter of the equalizer is obtained as a byproduct, when the signal $x_2(n)$ is filtered with the prototype filter upsampled by a factor of 2^8, or 256, as shown in Fig. 1. The resulting signal $x_1(n)$ does not require further processing, as it is the output signal $y_1(n)$ of the lowest band filter. The filter $H_{\mathrm{LP}}(z^{256})$ also implements the largest input-output delay, so a synchronization delay is unnecessary in the two lowest bands, as seen in Fig. 1.

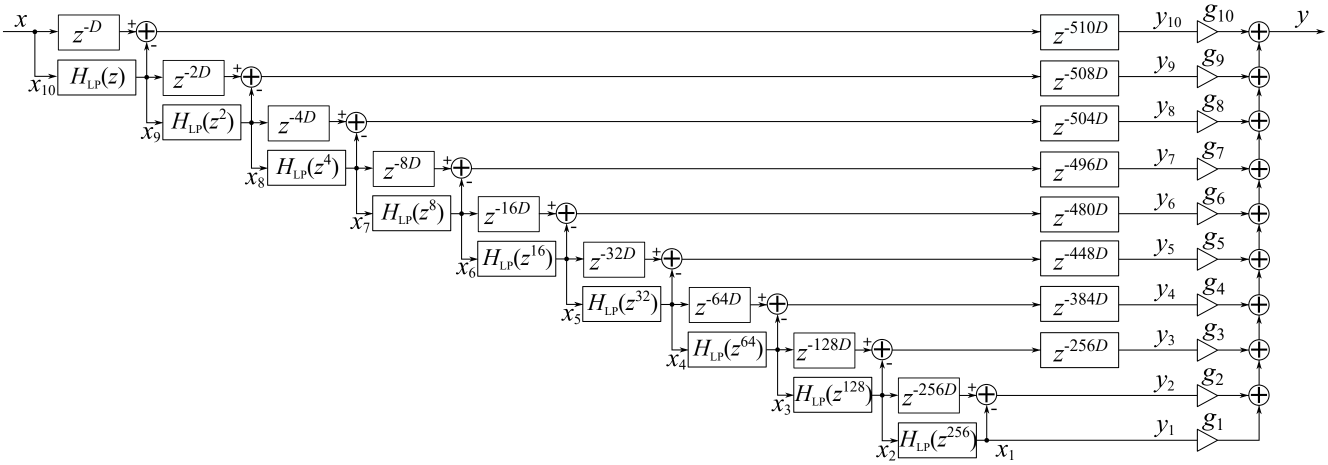


Figure 1: *Block diagram of the parallel FIR graphic equalizer for ten octave bands [4]. The signal path at the top produces the highest band (16 kHz) whereas the bottom one produces the lowest band (31.25 Hz).*

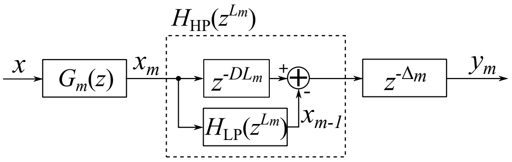


Figure 2: *Filters and delay lines associated with a single band for $m = 2, 3, ..., M$, cf. Fig. 1. This mth band transfer function $H_m(z)$ represents the relation between $Y_m(z)$ and $X(z)$.*

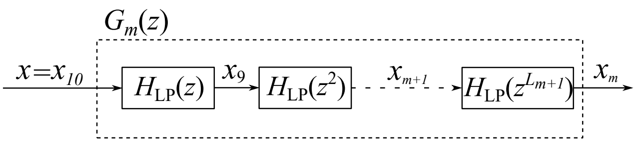


Figure 3: *Details of the transfer function $G_m(z)$ used in Fig. 2.*

Finally, as presented in Fig. 1, the desired gain factor g_m of each band is applied and the total response of the equalizer $y(n)$ is obtained as a weighted sum of all band output signals:

$$y(n) = \sum_{m=1}^{M} g_m y_m(n). \tag{8}$$

Since the band filters determine the gain on their own band very accurately, optimization of filter gains is unnecessary, and command gains can directly be used as weights g_m. It is worth remarking that the use of complementary filters guarantees a completely flat total response, ensuring the perfect reconstruction, as demonstrated in [4].

The performance of the described GEQ has been evaluated in [4] in terms of computational complexity, latency, and error. The total number of operations of the GEQ is 172, of which 64 multiplications and 108 additions. The latency, calculated following Eq. (7), is equal to 4599 samples, i.e., 95.8 ms. In [4], the proposed equalizer has been compared to other linear-phase approaches, i.e., single FIR GEQ and FFT-based GEQ, and the results are shown in Tab. 1. The single FIR GEQ has the same latency as the proposed method, but a much higher computational cost. The FFT-based approach shows the largest latency introduced by the frame-based FFT processing. Although the latency can be reduced by applying the zero-latency partitioned convolution [38, 39], the computational complexity remains higher than the proposed GEQ. Finally, the error is computed as the maximum magnitude difference between the desired and the obtained gains at the command frequencies of the GEQ, taking into account all the possible configurations with ±12 dB [40], and it is equal to 0.79 dB.

Table 1: Performance of the proposed equalizer compared with other linear-phase GEQs [4]. The best result in each column is highlighted.

Method	Latency	Mul	Add
Single FIR	**4 599**	4 600	9 198
FFT-based	20 983	116	168
Proposed	**4 599**	**64**	**108**

3. REAL-TIME IMPLEMENTATION

The proposed GEQ has been implemented as a PlugIn of the NU-Tech software [41], that is a platform specifically developed to test and tune real-time DSP algorithms through a PC workbench. The developer can write his own plugIns, called NUTSs (NU-Tech Satellites), in C++ and plug them into the GUI to test the results on a common PC. The internal parameters of every plugIn can be adjusted using the RTWatch (RealTime Watch).

Fig. 4 shows the flowchart of the real-time implementation of the IFIR GEQ, and Fig. 5 shows the NU-Tech interface used for experimental tests. The proposed GEQ is implemented in the NUTS “IFIR_GEQ” built as a standard C++ dll file. The RTWatch, shown at the bottom of Fig. 5, allows for the setting of twelve parameters, i.e., “ResetGdb,” which reset all the gains to zero, “Bypass,” which can bypass the equalizer and copy the input directly to the output with no elaboration, and the gains of the ten bands of the equalizer. When the user presses the play button of the transport panel, the instructions reported in Fig. 4 are executed. So, the

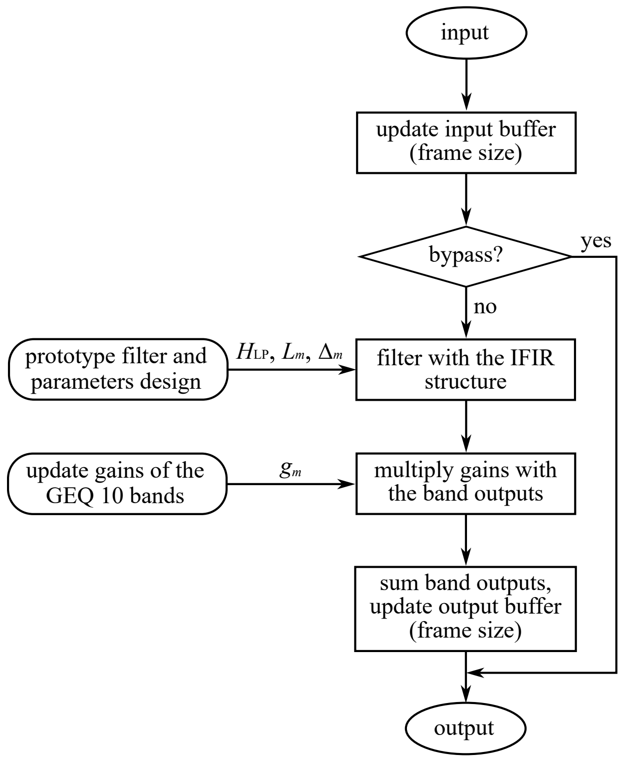


Figure 4: *Flowchart of the real-time implementation of the IFIR GEQ.*

input buffer, with a frame size length, is updated with input signal samples. If the bypass is selected, the input samples are directly copied to the output signal without elaboration. If the bypass is not active, the input signal is filtered with the IFIR structure, derived from the prototype filter, as shown in Fig. 1.

The prototype filter design and the calculation of the parameters are implemented once before the audio playback and remain always the same during the reproduction. Then, each mth band output is multiplied by the respective mth gain g_m. The gains are chosen by the user using the RTWatch and may change during the reproduction. As an example, in the RTWatch of Fig. 5 (i.e., the table at the left bottom of the NU-Tech board) a ± 12 dB zig-zag setting gain is chosen. Finally, the band outputs are summed and saved in the output buffer, so the output signal is updated. For the experiments, a sample rate of 48 kHz and a frame size of 8192 samples have been used.

4. EXPERIMENTAL RESULTS

The proposed GEQ has already been objectively validated in [4], evaluating its performance, as reported in Sec. 2. In this work, subjective tests are performed in a listening room shown in Fig. 6, using two loudspeakers Hedd Type 20 MK2 spaced two meters from the listener. The loudspeakers have been connected to a sound card Roland Rubix22, managed by an HP Notebook laptop Intel(R) Core(TM) i7.

To assess the perception of the audio quality of equalized sound, listening tests following the ITU-R BS.1284-1 standard [42] were performed. The subjects involved in the experiment were 10 expert listeners, ranging in age from 26 to 53 years. The term “expert” refers to a person with a technical background in acoustics, working in the audio field, and who is experienced with subjective listening tests. Tests conducted with this type of listener provide a better and quicker indication of likely long-term results. However, to avoid bias in the results of the listening test, the listeners were not familiar with the study and the expected results. The test was designed as a comparative test between the equalized and unequalized track, used as a reference, and the bipolar discrete seven-grade scale, shown in Tab. 2, was used.

Each listener, performed a total of 16 trials considering four reference signals of different music genres, listed in Tab. 3, and four equalization curves derived from the equalization curves used in [43], i.e.,

- Bass Boost (BB) with gains [3.43 3.43 3.43 3 2.5 1.3 -1 -6 -6 -6];
- Treble Boost (TB) with gains [-6.25 -5.63 -4.38 -2 3 3 3 3 3 3];
- Midrange Dip (MD) with gains [6.25 3.43 1 -1 -2 -2.2 -2 -3 2 -1];
- Midrange Boost (MB) with gains [-6.25 -3.43 -1 1 2 2.2 2 3 -2 1].

The “Bass Boost” and the “Treble Boost” curves emphasize low and high frequencies, respectively. The “Midrange Dip” is originated from an 80-phon equivalent loudness contour (ELC) of the ISO-226 standard [44], by deriving the exact gain of the ELC curve at the corresponding ten command frequencies. An equal-loudness contour is a measure of sound pressure over the frequency spectrum, for which a listener perceives a constant loudness when presented with pure tones [43]. Finally, the “Midrange Boost” curve is obtained by changing the control gains of the “Midrange Dip” by sign.

The proposed IFIR GEQ has been compared with the filterbank equalizer (FBEq) of [32]. Fig. 7(a) shows the four equalization curves obtained with the IFIR GEQ, and Fig. 7(b) shows the ones obtained with the FBEq, using a prototype filter with a length of 8000 samples. It is worth noting that the FBEq filter has a much higher computational cost than the IFIR GEQ, requiring a total of more than 600,000 operations per sample. Moreover, even the latency of the FBEq is very high above 300 ms.

In each test, each subject was subjected to the evaluation, with respect to the reference track, of the same track equalized with the IFIR Graphic Equalizer (IFIR GEQ), of the same track equalized with the Filterbank Equalizer (FBEq), and of the Hidden Reference, i.e., the reference, unequalized track hidden within the test to authenticate the listener’s reliability.

All test tracks had a duration of twenty seconds, following the guidelines of [42]. Initially, for all tests, the reference track was played so that the listener became familiar with the original sound. After that, for each reference signal, the program sequence was presented in random order, and the listener did not know which equalization methodology was under test. In addition, the listener was allowed to listen to a track under examination several times to obtain a more rational and less instinctive judgment. To evaluate the performance of the IFIR GEQ and FBEq against the original track, listeners were asked to rate some of the attributes suggested by [42] according to the seven-grade comparison scale shown in Tab. 2. The following attributes have been evaluated:

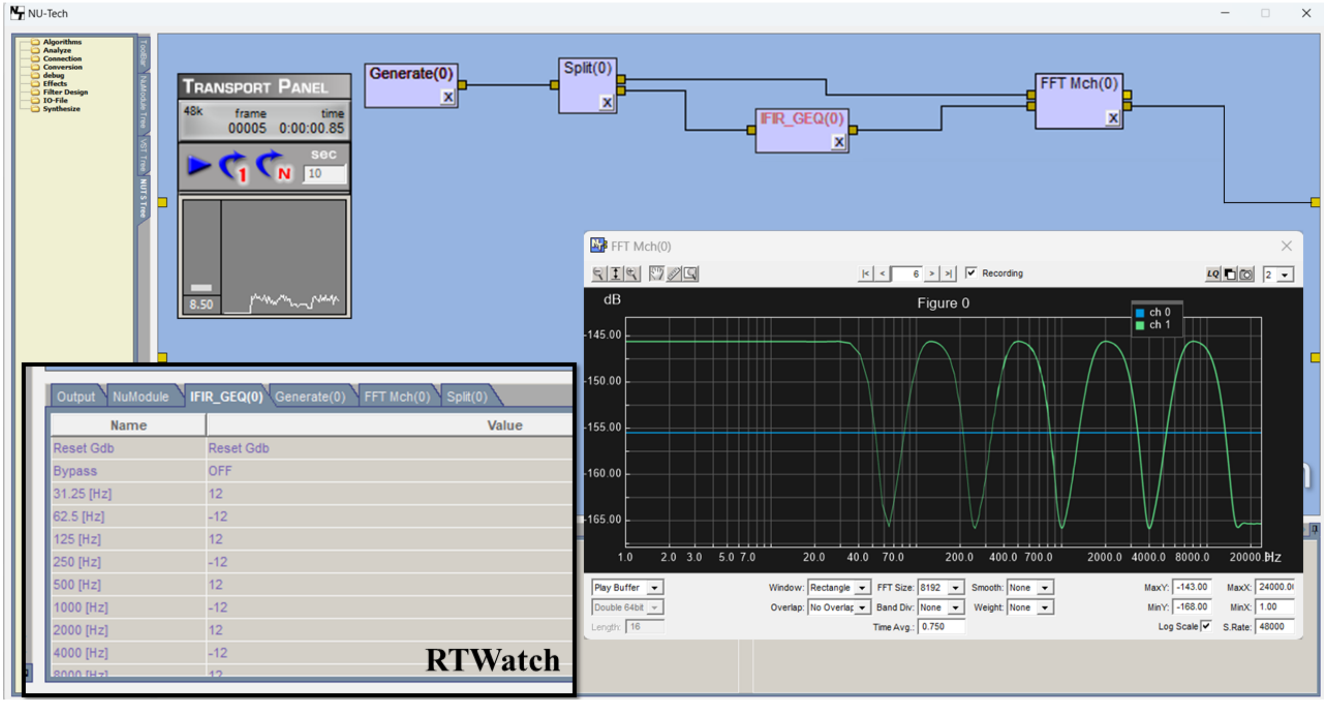


Figure 5: *Screen of the NU-Tech board used for experimental tests.*

Figure 6: *Photo of the setup used for listening tests.*

- Timbre, i.e., accurate portrayal of the different sound characteristics of sound source;
- Transparency, i.e., capacity to perceive all details of performance clearly;
- Main impression, i.e., the integrity of the total sound image, and the interaction between the various parameters.

The evaluation of the attributes is also affected by the nature of the type of equalization. However, although some attributes may present a rate lower than zero, which means that the original track sounds better than the equalized one, the two equalizers can be evaluated by comparing the score of each attribute.

The results of the subjective tests were processed by deriving the mean value and confidence intervals for the standard deviation, calculated considering a significance level of $\alpha = 0.05$, i.e., with a standard deviation (SD) of 95%, thus providing a statistically meaningful analysis.

Table 2: *Bipolar discrete seven-grade scale [42].*

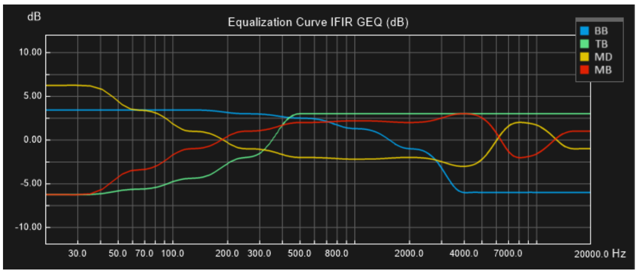
Comparison	
3	Much better
2	Better
1	Slightly better
0	The same
-1	Slightly worse
-2	Worse
-3	Much worse

Table 3: *Music tracks used for listening tests.*

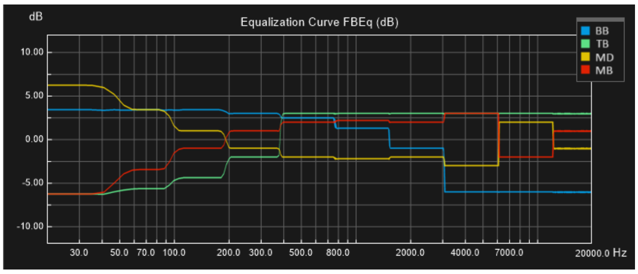
Genre	Author	Track
Classical	Tchaikovsky	Waltz of the Flowers
Jazz	H. Hancock	Edith and the Kingpin
Pop	G. Michael	Amazing
Rock	ZZ Top	Concrete and Steel

Tab. 4 reports the experimental results obtained with the equalization curve “Bass Boost” for all four music genres (i.e., classical, jazz, pop, and rock), comparing the IFIR GEQ with the FBEq. The analysis of the results of the “Bass Boost” equalization curve shows a very similar level of both algorithms for all attributes in each music genre. The most significant gap is in the transparency of the classical genre where the IFIR GEQ scored 0.33 points higher than its competitor equalizer at the same standard deviation. Overall, the FBEq was preferred five times versus the IFIR GEQ’s four times, and three times they scored the same.

Next, Tab. 5 shows the test results for the “Treble Boost” equalization curve for the four music genres. In this case, the results show a slight preference for the IFIR equalizer in that it



(a)



(b)

Figure 7: *Four equalization curves obtained (a) with the proposed IFIR GEQ and (b) with the FBEq.*

was preferred seven times out of twelve, and three times scored the same as the FBEq. In addition, although by only 0.11 points, the transparency of the IFIR GEQ in the pop genre was preferred to the reference track as opposed to the track equalized with the FBEq. In addition, for the pop and rock music genres, the IFIR GEQ was always preferred over the FBEq.

Tab. 6 shows the results obtained with the "Midrange Dip" equalization curve. This type of equalization curve improves all three attributes, in both the equalizers and with all four music genres. In particular, the IFIR GEQ scored better for timbre in each music genre. It also scored a higher main impression for jazz music than the FBEq by 0.44 points. The FBEq equalizer, on the other hand, was better in terms of transparency with the classical genre by 0.56 points. Overall, the IFIR GEQ was preferred six times against the FBEq's three times, and three times they scored the same.

Finally, Tab. 7 reports the results for the "Midrange Boost" equalization curve. In this case, there is a preference for the timbre with IFIR GEQ for each musical genre. In addition, the main impression for the pop genre of the implemented equalizer is found to be 0.5 points higher than that of the FBEq track. Also, as with the "Treble Boost" curve, the IFIR GEQ was preferred over its competitor for all attributes in the pop and rock music genres. In general, the IFIR GEQ was preferred nine times out of twelve, twice they got the same score, and only once was the FBEq preferred.

Overall, there is a slight preference for the IFIR equalizer in that it was preferred twenty-six times compared to eleven for the FBEq, and eleven times they scored the same. However, the main advantage of this graphical equalizer lies in the fact that the computational cost of making it is significantly lower than that required to make the Filterbank Equalizer. The use of the prototype FIR half-band interpolated filter and its stretched versions makes it possible to avoid multiplications with the null values of the filter during the filtering phase, and the symmetry of the prototype filter response makes it possible to halve the number of operations. In contrast, the FBEq with a subband structure, based on two anal-

Table 4: *Experimental results obtained with the equalization curve "Bass Boost". For each attribute and each music genre, the highest mean value is highlighted.*

Equalizer	Timbre		Transparency		Main impression		Genre
	Mean value	SD (95%)	Mean value	SD (95%)	Mean value	SD (95%)	
IFIR GEQ	-1.0	0.57	**-0.67**	0.65	-1.0	0.57	Classical
FBEq	**-0.89**	0.83	-1.0	0.65	-1.0	0.80	
IFIR GEQ	**-1.0**	0.92	-1.22	0.79	**-1.11**	0.95	Jazz
FBEq	-1.22	0.71	**-1.11**	0.76	-1.22	0.71	
IFIR GEQ	-1.5	0.33	**-0.9**	0.46	-1.5	0.44	Pop
FBEq	**-1.4**	0.32	-1.0	0.51	-1.5	0.53	
IFIR GEQ	-1.4	0.32	-1.7	0.42	-1.7	0.42	Rock
FBEq	-1.4	0.32	**-1.5**	0.53	**-1.6**	0.43	

Table 5: *Experimental results obtained with the equalization curve "Treble Boost". For each attribute and each music genre, the highest mean value is highlighted.*

Equalizer	Timbre		Transparency		Main impression		Genre
	Mean value	SD (95%)	Mean value	SD (95%)	Mean value	SD (95%)	
IFIR GEQ	**-0.5**	0.74	-0.38	0.63	-0.38	0.82	Classical
FBEq	-0.75	0.89	**-0.25**	0.72	-0.38	0.82	
IFIR GEQ	0.0	0.83	0.38	0.9	0.13	0.78	Jazz
FBEq	**0.25**	0.96	0.38	1.04	0.13	0.86	
IFIR GEQ	**-0.56**	0.87	**0.11**	0.83	**-0.78**	0.91	Pop
FBEq	-0.66	0.73	-0.33	0.73	-1.0	0.65	
IFIR GEQ	**-0.7**	0.78	**-0.5**	0.67	**-0.8**	0.87	Rock
FBEq	-1.1	0.74	-0.8	0.49	-1.1	0.68	

Table 6: *Experimental results obtained with the equalization curve "Midrange Dip". For each attribute and each music genre, the highest mean value is highlighted.*

Equalizer	Timbre		Transparency		Main impression		Genre
	Mean value	SD (95%)	Mean value	SD (95%)	Mean value	SD (95%)	
IFIR GEQ	**0.67**	0.57	0.44	0.58	0.67	0.57	Classical
FBEq	0.56	0.47	**1.0**	0.46	0.67	0.80	
IFIR GEQ	**0.56**	0.66	0.44	0.66	**0.55**	0.66	Jazz
FBEq	0.22	0.71	0.44	0.47	0.11	0.61	
IFIR GEQ	**0.89**	0.69	**0.89**	0.51	0.67	0.57	Pop
FBEq	0.67	0.57	0.67	0.46	0.67	0.57	
IFIR GEQ	**0.8**	0.64	0.9	0.46	0.9	0.54	Rock
FBEq	0.7	0.42	**1.0**	0.41	**1.0**	0.29	

Table 7: *Experimental results obtained with the equalization curve "Midrange Boost". For each attribute and each music genre, the highest mean value is highlighted.*

Equalizer	Timbre		Transparency		Main impression		Genre
	Mean value	SD (95%)	Mean value	SD (95%)	Mean value	SD (95%)	
IFIR GEQ	**-0.44**	0.58	**-0.11**	0.61	-0.78	0.71	Classical
FBEq	-0.78	0.44	-0.33	0.57	-0.78	0.71	
IFIR GEQ	**-0.3**	0.93	0.0	0.65	-0.3	0.78	Jazz
FBEq	-0.4	0.78	**0.1**	0.62	-0.3	0.78	
IFIR GEQ	**-0.6**	0.67	**-0.4**	0.6	**-0.5**	0.67	Pop
FBEq	-1.0	0.51	-0.6	0.6	-1.0	0.65	
IFIR GEQ	**-0.44**	0.66	**-0.33**	0.57	**-0.56**	0.66	Rock
FBEq	-0.67	0.33	-0.56	0.34	-0.89	0.22	

ysis and two synthesis filters for each direct branch to which are added two contributions due to the cross analysis and synthesis filters that take into account the interaction between adjacent bands, has a higher computational complexity.

5. CONCLUSIONS

The paper proposes a real-time implementation of a linear-phase octave-band graphic equalizer based on interpolated FIR filters. The GEQ is realized as a PlugIn inside the NU-Tech software. A first validation of the system has proven the performance of the GEQ, confirming its effectiveness in terms of error, latency, and computational complexity. Successively, the equalizer has been tested through subjective tests and compared with a filterbank equalizer. The experimental results have shown that the proposed equalizer is preferred in most of the cases and it presents a much lower computational load than the filterbank. Future plan of the system include the development of the IFIR GEQ on an embedded system.

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