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Filter Based Impedance Tube for Selection of Technical Textile Materials

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Abstract
Measurement of the sound absorption coefficient using an impedance tube is prone to errors due to various reasons. Conventionally minimising the errors requires additional hardware or proper calibration of many components used for measurement. This paper proposes a new error correction mechanism using a filtering technique. A low cost impedance tube was designed, developed and its performance was compared with a commercially available high cost tube. The errors which arose during the measurement of the sound absorption coefficient were minimised using different types of filters (low pass, band pass, LMS filtering). The purpose of the filter is to eliminate unwanted signals or noise which occur during the processing of analogue signals in the experiment. The system proposed was tested using various porous and non-porous functional textile materials and the results validated. A significant reduction in error was obtained at all frequency ranges with digital filters.

Key words: *impedance tube, low pass filter, band pass filter, LMS filter, noise reduction.*

Introduction

Measurement of the sound absorption coefficient is considered to be important in determining the acoustic properties of materials considered for use in noise control. The sound absorption coefficient is a quantity that represents the percentage of sound absorbed by a material. This is measured using the phenomenon of the reflection of sound waves. Sound waves are generated within a medium and transmitted towards the test sample. By measuring the incident and reflected waves, the reflection coefficient and, hence, the acoustic impedance can be calculated. The two standard methods for determining the absorption coefficient are the Standing Wave Ratio (SWR) method and transfer function method. The standing wave pattern and its pressure measurement are used in the former, whereas in the latter, the transfer function is used. The acoustic impedance is generally calculated over a wide range of frequencies, and these standard methods introduce errors in the measurement setup. Signals received from nature and those in various engineering applications usually consist of acoustic background noise. As a result the signals have to be cleaned up with digital tools before they produce output [1-2]. Noise reduction algorithms play a major role in eliminating those noises which are present in the signals. Various noise reduction algorithms have been proposed in the literature [3-4]. The noise reduction process can be achieved in many ways for different fields, such as signal/image de-noising [5], image enhancement [6], speech processing [7], mechanical fault diagnosis [8], biological data processing [9], etc. When designing a system, one of the most difficult parts is to carry out high quality filtering of unwanted information. In practice, it is not possible to completely eliminate the unwanted information. Hence advances in technology are used to diminish the noise coming from sources down to noise levels at which their negative effect on the important signal is negligible. Recent applications of several advanced filtering techniques have shown that the signal-to-noise ratio (SNR) can be increased by using appropriate filtering techniques [10-15]. All these techniques require prior knowledge of the noise, typically the noise variance or power spectral density, or the instantaneous signal-to-noise ratio (SNR), at all times [16].

In this paper, a novel approach for the reduction of noise is presented. Instead of making modifications to existing hardware, a digital technique was used. Three types of digital filters, namely low pass, band pass and LMS filters, were investigated. The filters were designed in the MATLAB environment, and the sound wave from the impedance tube was integrated with the system. The signals from the tube were then processed with the digital filters to reduce noise, and the sound absorption coefficient was estimated.

Sound absorption theory

Noise control is essential in vehicles that carry passengers over long distances, as they are confined to the same closed environment for a long period of time. Automotive companies employ various noise control materials at different locations. Generally in a car, materials like polyester, nylon, composites of carbon and aramid fibres are used [17]. The materials are mainly placed in interior fitments, safety facilities, tyre reinforcement, and carpets. They are also used for sound and thermal insulation.

Sound absorbing materials are used to soften the acoustic environment of a closed volume by reducing the amplitude of the reflected waves. Absorptive materials are resistive in nature, either fibrous, porous or, in rather special cases, reactive resonators, and provide some degree of absorption at nearly all frequencies (up to 2kHz). The absorption ratio and transmission loss, which represent the sound reflection and penetrating capability of a sample material, are measured by an impedance tube. The performance of sound absorbing materials is evaluated by the sound absorption coefficient (α). The sound propagation in a standing wave duct is assumed as stationary plane waves with zero mean flow speed, propagating in air. The complex acoustic pressure p(x,t) and particle velocity v(x,t) of the medium are given in ***Equations (1)*** and ***(2)*** [18]

$$p(x,t) = Ae^{j(\omega t - kx)} + Be^{j(\omega t + kx)} \quad (1)$$

$$v(x,t) = \frac{1}{z_0}\left[Ae^{j(\omega t - kx)} - Be^{j(\omega t + kx)}\right] \quad (2)$$

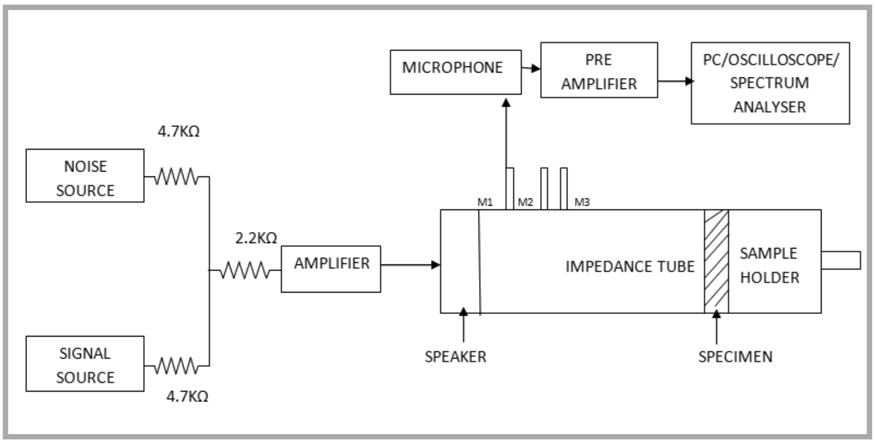

***Figure 1.** Block diagram of tube designed.*

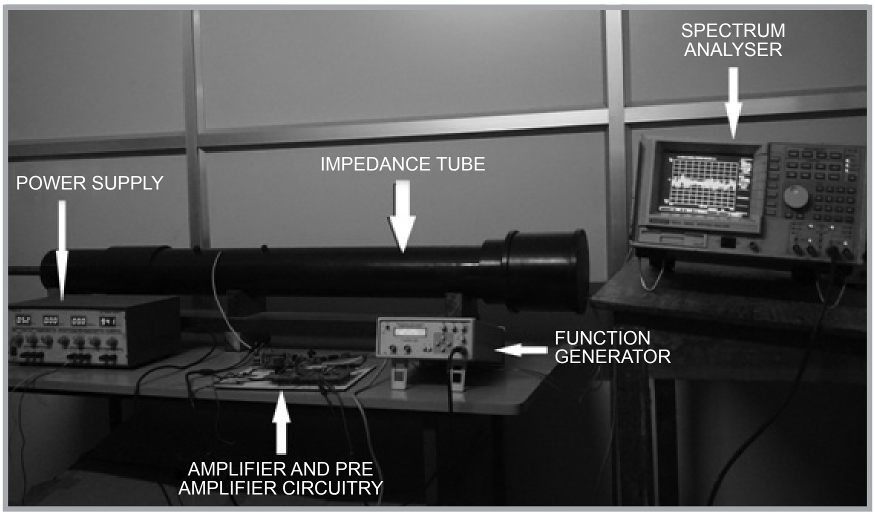

***Figure 2.** Impedance tube setup.*

where A and B are amplitudes of incidence and reflected waves, respectively, ω the angular frequency (rad/s), K the wave number, z_0 the characteristic impedance (Ω) of air at 20 °C, and ρ (kg/m^3) and c (m/s) are the air density and speed of sound in air, respectively. The low cost impedance tube is designed as shown in ***Figure 1***.

The impedance tube set up contains a signal source along with power amplifier section. The signal characteristics of the existing impedance tube are improved by modifying the signal source from generating monotone signals into one that generates white noise signals. The main purpose of adding white noise is to provide a uniform reference frame in the time-frequency space. Also a signal embedded with white noise permits a frequency analysis resolution of 6.25 Hz in a frequency domain of 125-10000 Hz.

Experimental setup

To test the sound absorption coefficient, a transfer function based impedance tube was developed. One microphone method was employed by using a single microphone at two locations successively, thereby avoiding phase mismatch between two different microphones [19]. The theory of transfer function method has already been discussed extensively in the literature [20]. The impedance tube developed along with the entire setup is shown in ***Figure 2***. The inner diameter of the tube for this design is d = 104.8 mm, which gives an upper limiting frequency of f_u = 1.8 kHz. The spacing between the sound source and microphone is taken as x = 720 mm, and the total tube length is 1420 mm. The impedance tube setup contains a signal source along with a power amplifier section. The noise circuitry is designed by adding the signals generated from two similar signal generators through a resistive network. The signal thus generated is amplified using a power amplifier circuit and fed into the impedance tube through a loud speaker.

The microphone that is placed inside the tube captures the incident signal and reflected signal from the sample. These signals are too weak to be transmitted to recording devices, and hence preamplifiers are used to increase the microphone signal to a line-level by providing stable gain, while preventing induced noise that would otherwise distort the signal. The preamplifier is placed close to the microphone to reduce the effects of noise and interference. The amplitude of the incident and reflected signals vary according to the sound absorption of the sample. Therefore by monitoring the amplitude variations of both signals and taking the ratio, the sound absorption coefficient of the material is calculated. A digital storage oscilloscope, spectrum analyser and PC are used for storing and analysing these signals.

Filtering

The signal acquired may contain noise which might be due to ambient sources, electromagnetic interference, and faults in the electric circuitry during acquisition. Some noise such as a power frequency hum can be easily identified and filtered, whereas others such as human and animal noises are harder to classify and filter. In this paper, three types of digital filters were designed and tested with various samples using an impedance tube.

Low pass filtering

If most of the noise added to the signal is due to reflections and harmonics, then a simple low pass filter is sufficient to mitigate the noise in the data. The low pass filter is designed with a 3 dB stop frequency at a slightly higher frequency than the original signal frequency. The filter attenuates frequency components higher than the cutoff frequency but maintains the original signal for frequency components less than the cutoff frequency. Two-stage RC circuits are frequently used as low-pass filters. Their frequency response can be derived by standard circuit analysis techniques. For a sinusoidal input voltage V_i (V), the output voltage from one stage is

$$V_o = V_i \frac{Z_c}{Z_c + R} \quad (3)$$

where, $ZC = 1/i\omega C$ (Ω). In practice only the amplitude ratio is required, which is

$$\frac{V_o}{V_i} = \frac{1}{\sqrt{[1+(\omega\tau)^2]}} \quad (4)$$

where $\tau = RC$ is the time constant of the circuit.

When two RC circuits are connected in series with a buffer amplifier, the output voltage of the first becomes the input to the second. The final output is therefore just a product of factors, as in ***Equation (5)***,

$$\frac{V_o}{V_i} = \frac{1}{[1+(\omega\tau)^2]} \quad (5)$$

The low frequency response is essentially the same as for the single RC circuit, but the attenuation falls off more rapidly with increasing frequency above 0. The time response of the filter can be found by setting up and solving differential equations for the circuit. For a single-stage RC with an input voltage going from zero to V_i at $t = 0$, the output voltage is:

$$V_o = V_i\left(1 - e^{\frac{-t}{\tau}}\right) \quad (6)$$

The corresponding result for the two-stage RC is:

$$V_o = V_i\left(1 - \left(\frac{t}{\tau} + 1\right)e^{\frac{-t}{\tau}}\right) \quad (7)$$

The low pass filter is modelled with the fdesign function in MATLAB. The filtering process is done for different sample materials and compared with the system without filters.

Band pass filtering

Band pass filters allow only certain frequency ranges and attenuate the components of frequency higher and lower than the current frequency. Since our input source signal is a pure sine wave generated by a signal generator, band pass filters are most ideal for filtering noise in the system. The band pass filter is designed by using the fdesign function in MATLAB with the sample rate, and frequencies for the stop band and pass band as parameters.

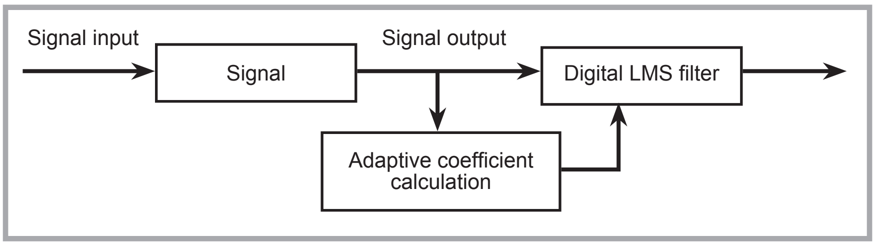

Figure 3. *Block diagram of the noise reduction method proposed.*

Table 1. *Properties of samples.*

Material property	Glass wool	Polyester	Comber
Mass per unit area, g/m^2	800	400	500
Thickness, mm	12.00	9.00	11.00
Diameter, mm	90	90	90
Porosity, %	55	75	65
Air permeability, cm^3/cm^2/s	454	351	382
Fibre shape	Nearly round	Round	Tubular shape

LMS filtering

The noise reduction problem considered in this paper is to recover a signal of interest (SOI) x(n) from an observation signal y(n) which is corrupted by noise v(n), as in ***Equation (8)***.

$$y(n) = x(n) + y(n) \quad (8)$$

where v(n) is the additive noise, which is a Gaussian random process. It is assumed that the noise v(n) is uncorrelated with the SOI x(n) signal. By applying the Discrete Fourier Transform (DFT) to ***Equation (8)***, the relationship of the signal model in the discrete frequency domain is obtained. The least-mean-square (LMS) filter uses adaptive filtering to mimic the performance of another filter specified. Considering an N-tap filter, with the weight vector w(n) at time instant T,

$$w(n) = [w1(n)\ w2(n)\ \dots\ wN(n)]T \quad (9)$$

Let {x(n)} be the input sequence and x(n) = [x(n) x(n-1) … x(n – N + 1)]T be its vector representation containing the immediate past N samples of {x(n)}. The filter output y(n) = wT (n)x(n) aims to follow a desired signal, and the estimation error e(n) is defined by

$$e(n) = d(n) - y(n) \quad (10)$$

An adaptive filtering algorithm adjusts the filter tap weight w(n) at each time instant according to the value of e(n) measured, as in ***Equation (11)***.

$$w(n+1) = w(n) + \mu e(n)x(n) \quad (11)$$

where μ is defined as the step-size parameter which affects the convergence behaviour of the filter weights. A block diagram of the noise reduction method is shown in ***Figure 3***. The unprocessed noisy signal is segmented every 40 ms in order to achieve effective noise reduction.

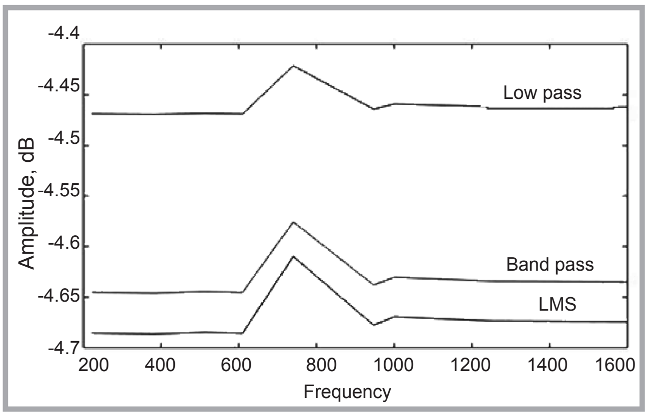

Figure 4. *Response of filters for glass wool data.*

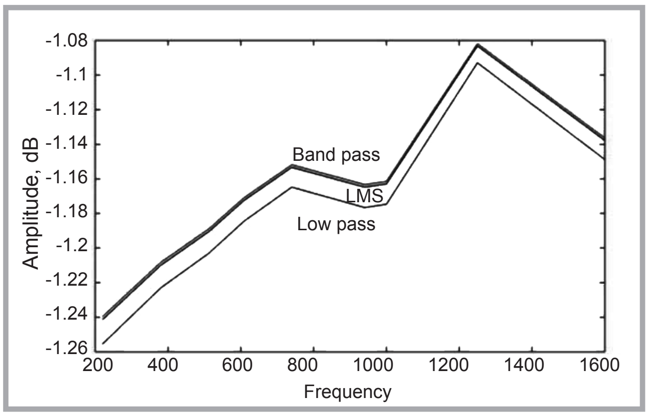

Figure 5. *Filter response for polyester data.*

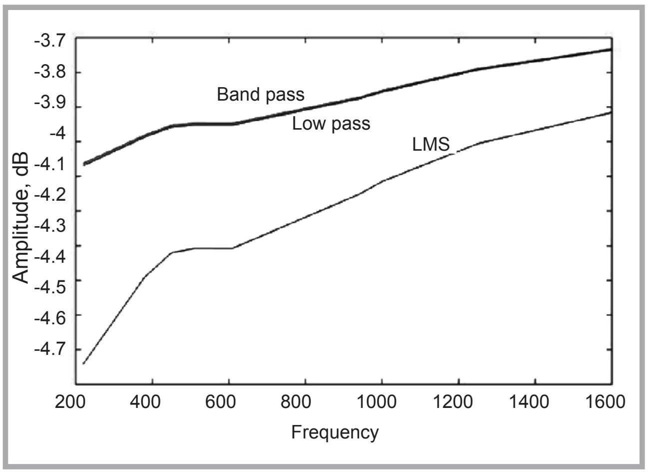

Figure 6. *Filters response for comber data.*

In the case of comber (***Figure 6***), low pass filter and band pass filters showed a similar response, suggesting the lack of higher frequency noise components in the signal. But LMS filter shows higher attenuation than the other two filters. From the observations, it can be suggested that band pass filtering will be more reliable for noise filtering as the signal is a pure sine wave without harmonics.

The filtered data for different samples is compared for the sound absorption nature. The samples showed nearly a flat frequency response. It is observed that glass wool has a higher sound absorbing nature when compared to the other two samples. The unfiltered data is less consistent when compared to filtered data, and hence is more reliable for data analysis. A comparison of unfiltered and filtered data is given in ***Figures 7*** and ***8*** and their actual values shown in ***Table 2***.

In ***Table 2***, original data refers to the readings obtained from the standard B&K tube, while unfiltered data refers to those of the impedance tube developed without filters, and the filtered data represents the values after applying filters. It can be inferred that the impedance tube designed provides an accuracy of 74% (error-24%) for glass wool on average. With the inclusion of an LMS filter, the accuracy increases to 94% (error-6%). Similarly for polyester and comber, the accuracy of the impedance tube is 73%

In the method proposed, the problem of noise cancellation is stated as the process of identifying a matching weight vector wt for each noisy frame yt, since a whole unit with several frames when treated as a segment of the consecutive speech frame can be identified more accurately from noise than the whole frames. The LMS filter in this case is designed to mimic the output from the bandpass filter.

Results and discussion

The filters were tested for different samples and the consistency of the results analysed. Three materials – glass wool – 800 GSM/40 mm, polyester – 400 GSM/40 mm and comber – 500 GSM/40 mm were used for studying the effect of filters in reducing the errors in the impedance tube. Properties of the materials used are given in ***Table 1***.

Figure 4 shows that more signal is attenuated for band pass and LMs filters. The signal response of LMS and band pass filters are identical.

The similarity of responses between LMS and band pass filters are much evident in this case (***Figure 5***). However, the low pass filter shows reduced amplitude when compared to other filters, suggesting more harmonics and fizzle noise present in the data. Also the amplitude of the signal is higher when compared to the glass wool data, suggesting a lesser absorbing nature of the material.

Table 2. *Absorption coefficient (α) for different samples.*

	Glass wool			Polyester			Comber		
F (Hz)	original	unfiltered	filtered	original	unfiltered	filtered	original	unfiltered	filtered
16	0.034	0.036	0.030	0.363	0.369	0.360	0.115	0.119	0.113
20	0.030	0.037	0.030	0.263	0.266	0.260	0.028	0.033	0.026
25	0.043	0.040	0.029	0.187	0.193	0.183	0.027	0.031	0.026
32	0.016	0.042	0.029	0.170	0.177	0.168	0.034	0.038	0.033
40	0.022	0.036	0.030	0.062	0.066	0.059	0.026	0.029	0.024
50	0.020	0.029	0.030	0.074	0.077	0.070	0.031	0.033	0.029
63	0.021	0.034	0.0309	0.051	0.055	0.045	0.035	0.039	0.031
80	0.033	0.042	0.032	0.034	0.038	0.030	0.024	0.028	0.022
100	0.030	0.050	0.034	0.020	0.022	0.016	0.040	0.044	0.038
125	0.037	0.061	0.036	0.015	0.019	0.012	0.047	0.051	0.042
160	0.053	0.075	0.049	0.004	0.009	0.003	0.063	0.066	0.060
200	0.069	0.062	0.057	0.002	0.006	0.002	0.083	0.088	0.080
250	0.089	0.091	0.077	0.009	0.013	0.006	0.096	0.099	0.093
315	0.127	0.129	0.126	0.013	0.017	0.011	0.134	0.138	0.132
400	0.176	0.190	0.170	0.019	0.023	0.014	0.199	0.202	0.197
500	0.229	0.211	0.221	0.021	0.026	0.018	0.301	0.303	0.299
630	0.302	0.296	0.293	0.027	0.031	0.021	0.454	0.458	0.452
800	0.390	0.375	0.383	0.031	0.034	0.026	0.686	0.689	0.682
1000	0.488	0.472	0.437	0.043	0.046	0.039	0.885	0.889	0.882
1250	0.607	0.619	0.610	0.047	0.051	0.043	0.940	0.946	0.936
1600	0.745	0.757	0.741	0.068	0.073	0.062	0.799	0.803	0.798

and 94%, respectively. With the addition of filters, it increased to 88% and 96% for polyester and comber, respectively. Overall the accuracy of the impedance tube was increased by 12% with the LMS filter. Thus the reduction of noise in the signal measured improves the accuracy of the impedance tube.

Conslusions

The reduction of noise in any measuring instrument is important as it eliminates the error due to noise and increases the accuracy of measurement. An impedance tube was developed to measure the acoustic properties of functional textile materials. The values obtained from the impedance tube developed deviated from the ones obtained from the standard B&K tube due to various errors. To eliminate the error, a noise reduction method was proposed using various types of digital filters including low pass, band pass and LMS filtering. The filters were designed using the MATLAB environment and the results analyzed using three types of textile materials. LMS filtering was seen to perform better in terms of noise reduction and provides accurate results on par with the standard B&K tube.

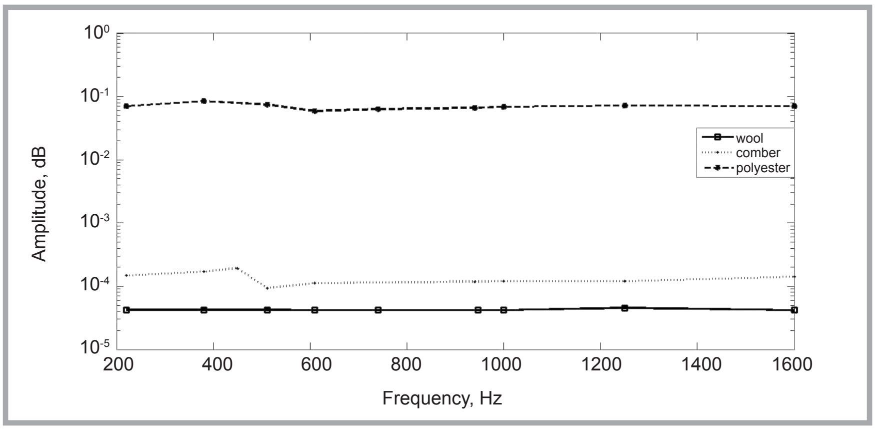

Figure 7. *Comparison of unfiltered data.*

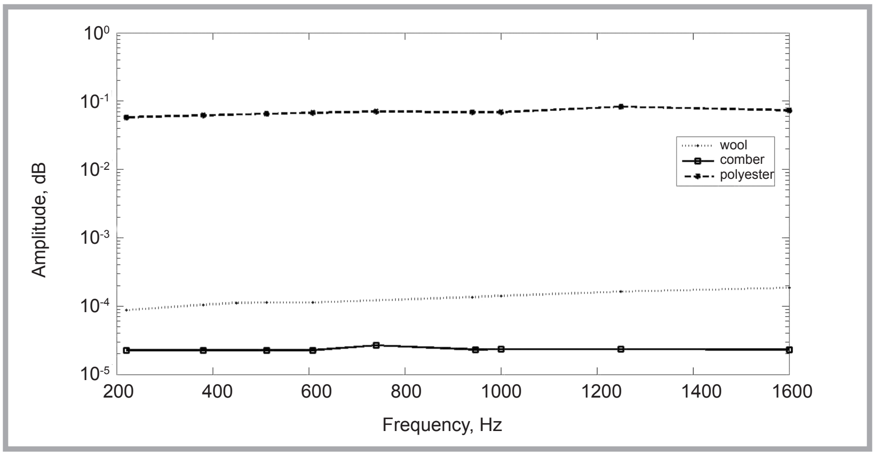

Figure 8. *Comparison of filtered data.*

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