Computer implementation of air absorption phenomena on a propagating sound wave

As a digital FIR filter to process signals in real time

Basis

The propagation of sound through air carries some energy dissipation, function of the frequency. Some of the factors [1] that influence the magnitude of that dissipation are:

- The non-uniformity of the propagation medium due to meteorological (refraction and turbulence). This will not be a main issue unless the volume of the enclosure is big enough to allow for considerable temperature changes and / or wind.
- Ground effect (it is an elastic medium capable of transmitting dilation and shear waves). This can be adequately modeled assuming that it is "locally reacting" [2], so the waves within are negligible and the ray-tracing will be enough to describe its contribution.
- Absorption of sound in air.

The last addend is of special interest when simulating room impulse responses, since the traveled distance after a few reflections will effectively introduce a considerable attenuation. The lower bands of the audio frequencies are left almost untouched, while the higher ones (specially above 2 – 4 kHz) are highly attenuated. This attenuation is not linear, but exponential with the distance, so it can be described with a logarithmic unit such as [dB / km]. This parameter is denominated "atmosphere attenuation coefficient", and is a function of the frequency of the propagated signal.

The mechanism by which the acoustical energy is absorbed in the air are mainly two [3]:

- classical absorption: viscous losses due to friction between air molecules which results in heat generation
- relaxational processes: sound energy is momentarily absorbed in the air molecules that can then re-radiate sound at a later instant and partially interfere with the incoming sound

The values of the attenuation coefficient can be calculated theoretically [4] and obtained empirically [5]. A set of values assumed to be applicable for most circumstances can be found in the standard ISO9613-2 [6]. The latter come as a set of discrete values corresponding to octave bands from 63 to 8000 Hz.

Procedure

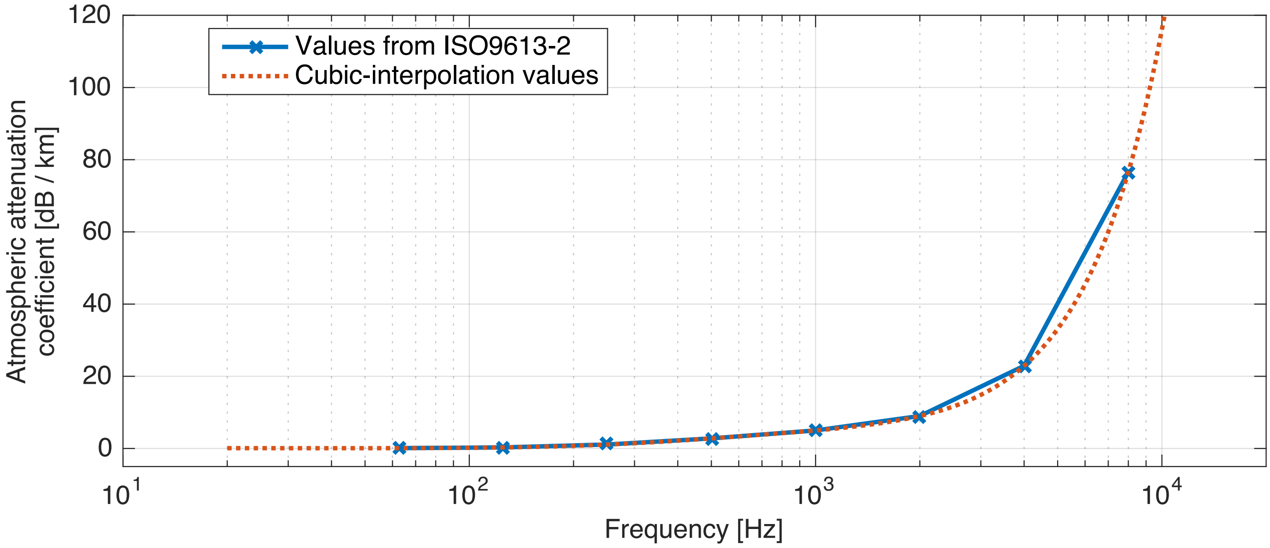
Based on the octave band values provided by the ISO9613-2, a digital filter is generated that can be used to process signals.

The desired impulse response of a FIR filter that emulates the behavior of air absorption is constructed from the frequency response using the inverse Fourier transform in its discrete form.

$$h[n] = F^{-1}\{H(f)\}$$

The frequency response used must be uniformly sampled in frequency. As the data provided by [**ISO9613-2**] is not, it needs to be resampled.

Using the data from table 2 on the ISO9613-2 and performing a cubic interpolation [7], the attenuation coefficient can be resampled. The result of doing so, with uniformly distributed frequency bins from 20 to 20000 Hz is shown here:



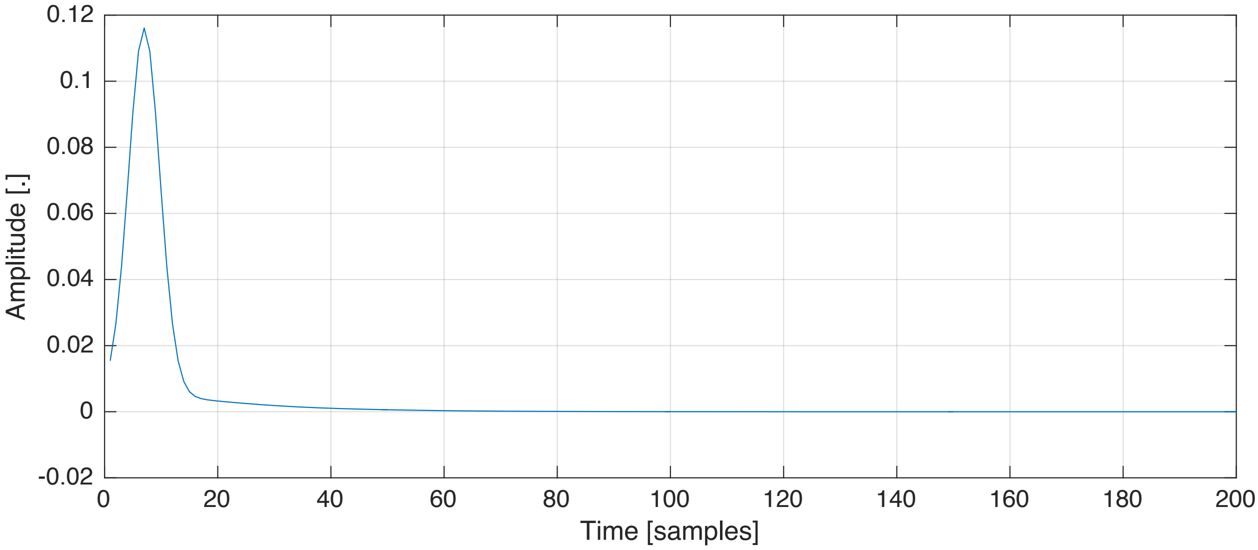
This frequency response can now be converted into an impulse response by using the inverse Fourier transform. Previously, the attenuation coefficients are converted to a linear scale by doing:

$$a_{abs}(f_k) = 10^{\frac{-A_{abs}(f_k)}{20} \cdot \frac{d}{1000}}$$

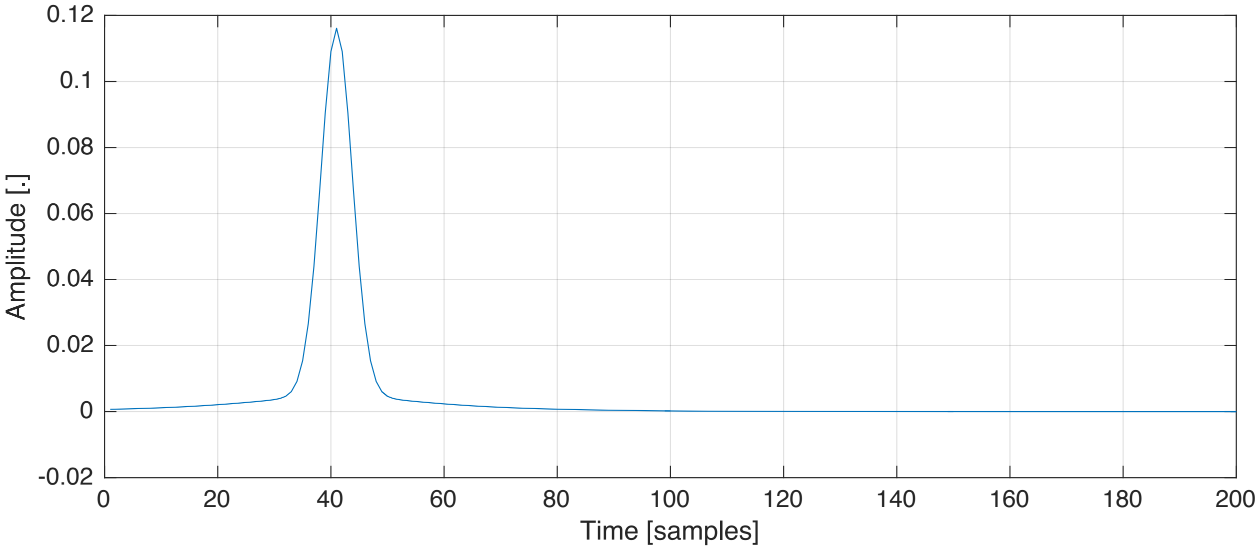
Being **A**$_{abs}$ the attenuation coefficient in [dB / km], **a**$_{abs}$ its counterpart on a linear scale [.] and **d** the distance to the source in kilometers. This dimensionless values **a**$_{abs}$ will be considered the module of the frequency response of the desired filter. The phase will be linear with non-zero slope, to produce a causal impulse response [8]. The slope of the phase will be set to a value that provides the lowest group delay, yet assures the obtained IR to be causal and stable (non oscillating).

$$|H(f_k)| = a_{abs}(f_k)$$
$$\angle H(f_k) = -\frac{f_k}{g\pi f_s}$$

Being **g** the group delay of the desired impulse response and **fs** the sampling frequency. Its value is selected such that the obtained impulse response fulfills previous conditions, while keeping it as low as possible. The minimum group delay of the filter will depend on the amount of attenuation the filter has to provide, consequently on the maximum distance to the source that has to be emulated. For example, in case the maximum distance is set to 500 meters (sampling frequency of 48000 Hz), giving the group delay a value of 8 samples will produce an impulse response that has “lost” some information at the beginning:



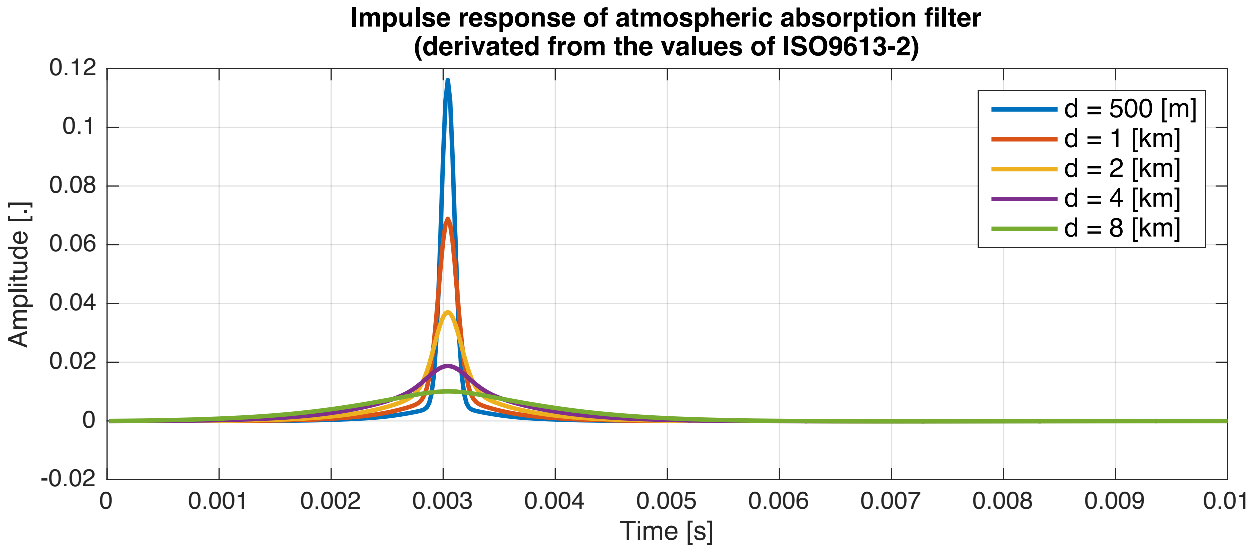
If the group delay is set to 40 samples, instead:



The consequence of have a non-zero group delay is the introduction of a small delay in the processed signal. This is not avoidable (for the filter to be causal) and has to be considered when evaluating the feasibility of the system.

Conclusion

The obtained impulse response has a Lorentzian shape [9] and necessarily introduces a non-zero group delay. The shape approximates a delta as the distance to the source tends to zero.



As seen on the last figure, when the distance grows, not only the peak gets smaller, the shape of the bell changes to a more flat one. This translates into having a variable frequency response, function of the distance to the source.

References

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