Computers & Fluids 64 (2012) 91–97

Contents lists available at SciVerse ScienceDirect

Computers & Fluids

journal homepage: www.elsevier.com/locate/compfluid

Development and validation of a phase shift and attenuation free filtering for exploitation of CFD data

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ARTICLE INFO

Article history:
Received 2 November 2011
Received in revised form 14 March 2012
Accepted 14 May 2012
Available online 23 May 2012

Keywords:
Signal processing
Aliasing
Filtering
Deconvolution
Attenuation
Phase shift

ABSTRACT

A causal filtering method is developed to perform decimation and storage of aerodynamic data without signal distortion. When done without precaution, decimation of instantaneous computed flow fields for storage and subsequent exploitation generates aliasing. Use of a low-pass filter limits the creation of spurious noise but attenuates signal amplitudes and shifts phases. The proposed filtering method combines the application of a low-pass filter before decimation, during the aerodynamic simulation, with that of a corrective filter, applied in the frequency domain after the full storage. This corrective filter is designed to correct attenuation and phase distortion caused by the low-pass filter. The method is validated on analytical signals and successfully applied to turbulent shear layer and noise radiation problems.

1. Introduction

The increase of the computational power observed during the last decades currently allow people to use numerical simulations as an efficient prediction and optimisation tool for aircraft design and conception in many topics such as combustion, turbine blade cooling, aerodynamic performances and noise emission, for instance. For noise reduction, a commonly used approach consists in simulating the flow field using Direct Numerical Simulation (DNS) or Large Eddy Simulation (LES) to compute the noise sources generated by the flow and its turbulence, as done by Freund [1] and Bogey and Bailly [2]. Such simulations nevertheless require dense computational grids to properly resolve turbulence and flow gradients, and a direct computation of the noise from generation zones to the far field in the same calculation is out of consideration. It is preferred, outside of the source zones and in the uniform medium, to perform either noise radiation using integral formulations such as Lighthill [3], Kirchhoff [4] or Ffowcs Williams and Hawkings [5], or sound propagation based on isentropic linearised Euler equations [6], which can provide the time signatures in the far field with a limited computational cost compared to the resolution of the Navier–Stokes equations. This one-way coupling is often performed using files for data exchange, leading to an important amount of stored data.

Frequency ranges of interest in aerodynamics and acoustics are different. In the acoustics, the low frequency part of the spectrum contains most of the energy and is especially investigated. Thus, storing every time step from the aerodynamics for noise radiation may not be relevant and can lead to highly over-resolved acoustic waves in the time domain, resulting in too long computations and a too large amount of data to store. This is especially visible when considering Lighthill integral formulation, where the five aerodynamic variables ρ, p, u_x, u_y, u_z are to be stored over the whole integration volume. Based on this consideration, the acoustic radiation step in usually performed using 1 every n time steps of the aerodynamic simulation. This decimation of the aerodynamic signals for their acoustic post-processing reduces the amount of stored data and numerical operations by a factor n, but without applying any filter before this operation aliasing occurs and frequencies above the cut off contaminate the resolved ones.

Such an approach is nevertheless successfully applied daily for aeroacoustics simulations using a surface integral method, where the flow variables are stored on surfaces surrounding the noise sources and on which turbulence is limited and low frequency acoustic fluctuations predominate (see simulations of Vuillot et al. [7,8] for instance). In that case, the grid used for the simulation is dense in the noise generation volume and progressively coarsened towards the storage surfaces; numerical dissipation hence plays the role of a low-pass filter that makes aliasing negligible during the decimation of the signal on the storage surface.

This approach can however not always be used when data are required to be stored in the volume. Indeed, inside the generation

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0045-7930/$ - see front matter
http://dx.doi.org/10.1016/j.compfluid.2012.05.008

domain the grid is dense to ensure a correct resolution of the turbulence and corresponding noise sources. Even not predominant, levels above the cut-off frequency thus contaminate lower frequency levels during the decimation process and lead to erroneous time signals, as observed by Perez et al. [9].

In this latter case, it becomes necessary to apply a low-pass filter to the aerodynamic fields, in the time domain, before their decimation and storage to limit aliasing. This filter being causal, it introduces attenuation and phase shift that modify the time signature of the considered signals. Considering for instance noise radiation using Lighthill analogy, attenuation can cause an underprediction of the radiated noise and phase shift a wrong evaluations of the non linear term $\rho u_i u_j$, which can finally lead to fully erroneous results.

Application of an inverse filtering, also referred as deconvolution or inverse convolution in the literature, is then required to restore the altered frequencies of interest, in terms of amplitude and phase. Such inverse convolutions of filtered signals have theoretically been successfully performed in the frame of geophysical data, without considering decimation. Depending on the signal to be treated, this deconvolution can be performed using either analog or digital techniques. The first technique is used for analog signals and requires the design of an analog inverse filter, as discussed by Burch et al. [10], which does not correspond to the case considered here. The digital technique, described for instance by George et al. [11], is well-suited for discrete signals and is based on the division of the altered signal by the low-pass filter in the frequency domain.

The objective of the work presented here is to apply a full filtering procedure to data representative of those to be stored during CFD simulations. This procedure is composed of a low-pass causal filtering and a decimation of the time signals followed by a digital deconvolution in the frequency domain. Influence of aliasing on signal distortion, which was not discussed by previous authors [10,11], will especially be investigated.

The paper is organised as follows. The filtering approach is presented in Section 2. Influence of aliasing on signal distortion and selection and characteristics of the low-pass filter are also detailed. Application to analytical signals and to realistic problems are then performed in Sections 3 and 4, respectively. Finally, conclusions are given in Section 5.

In the following, all represented power spectral densities (PSDs) are obtained using the method of averaged periodograms. The initial signal is decomposed into n_r distinct signals without overlapping. Each signal is periodised using the Hanning window before being Fourier transformed; levels amplitudes are then averaged over the n_r realisations to ensure the statistical convergence of the spectra.

2. Presentation of the filtering approach

2.1. Theoretical description

Because of the storage of one every n time step during the aerodynamic simulation, application of a low-pass filter before the decimation of the time signals is required to limit aliasing. This filter F_{lp} can be written in the following way, in the frequency domain:

$$F_{lp}(f) = A(f) \cdot e^{i\varphi(f)} \tag{1}$$

where A stands for the attenuation of the filter and φ to its phase shift. Following the results of George et al. [11] and considering that the filter leads to a negligible aliasing caused by the decimation of the signal, applying the corrective filter F_c defined by Eq. (2) permits to obtain a signal identical to the initial one, for frequencies below low-pass filter cut off.

$$F_c(f) = \frac{1}{A(f)} \cdot e^{-i\varphi(f)} \tag{2}$$

Practically, the use of this second filter is to be performed in the frequency domain and thus requires the whole signal to be stored before being Fourier transformed and the correction to be applied.

When dealing with non periodic data, the time signal is usually periodised with a window to suppress or reduce discontinuities at the beginning and end of the signal, thus limiting introduction of spurious high frequency noise. Such an operation is nevertheless not possible in the present case, as it becomes impossible to remove the contribution of the window to the signal once the corrective filter has been applied. Indeed, when noting s_f the low-pass filtered signal and H the periodic window (in the time domain), the windowed corrected signal s_{wc} is written:

$$s_{wc}[k] = ((s_f \cdot H) \star F_c)[k] \tag{3}$$

where $\star$ corresponds to the convolution product and k to the current time step. From this expression, it clearly appears that no simple procedure exists to remove the contribution of the window to the corrected signal.

2.2. Low-pass filter selection

The method presented hereabove is theoretically exact for periodic signals, when using a low-pass filter with infinite attenuation above the cut off, leading to no aliasing. Such a filter does not exist for practical use, for which only causal filter with continuous attenuation around the cut-off frequency can be used. Signal amplitude correction for frequencies close to the cut off therefore corresponds to amplifying spurious aliased noise, in addition to the signal that is to be conserved. To alleviate this problem it is possible, during the corrective filtering, to annihilate levels at frequencies close to the cut off, for which aliasing remains important and level amplification would lead to important spurious noise.

Determining the limit frequency for which aliasing can be considered negligible depends on the low-pass filter used before the decimation. Using $p[k]$ for the non filtered and $p_f[k]$ for the filtered signals, respectively, at time step k, any causal numerical filter can be written in the form:

$$p_f[k] = \sum_{i=0}^{N} b_i \cdot p[k-i] - \sum_{i=1}^{N} a_i \cdot p_f[k-i] \tag{4}$$

In that expression, N corresponds to the filter order and a_i and b_i to the filter coefficients. Using $a_i = 0$, filtered signal only depends to non filtered past values. This corresponds to Finite Impulse Response (FIR) filters, that are always stable in time. On the contrary, using coefficients $a_i \neq 0$ corresponds to Infinite Impulse Response (IIR) filters, that can cause the filtered signal to diverge but which present a better attenuation compared to FIR filters with the same order N. First order IIR filters indeed provide an attenuation of -20 dB/decade, compared to -10 dB/decade for same order FIR filter, for instance. Interested reader may refer to Oppenheim et al. [12] for more details about digital filters.

Considering that memory requirements might become problematic when the filtering is to be performed on a large amount of grid points and aerodynamic variables, use of a IIR filter here appears to be the most appropriate for efficient and manageable low-pass filtering. In the following, all investigations are thus performed considering a first order IIR filter.

2.3. Aliasing effect on signal distortion

In this subsection, the consequences of the aliasing due to the filtering and decimation of the initial signal are investigated to