STATE OF THE ART TECHNIQUES USED FOR NOISE SOURCE IDENTIFICATION ON COMPLEX BODIES

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Abstract

Over the last few decades, many approaches have been undertaken in order to asses detailed noise source identification on complex bodies, i.e. aircrafts, cars, machinery. Noise source identification implies to accurately obtain the position and frequency of the dominant noise sources. There are cases where traditional testing methods can not be applied at all or their use involves some limitations. Optical systems used for near field analysis require a line of sight that may not be available. The state-of-the-art technology for this purpose is the use of a large number of microphones whose signals are acquired simultaneously, i.e. microphone phased array. Due to the excessive cost of the instruments and the data acquisition system required, the implementation of this technology was restricted to governmental agencies (NASA, DLR) and big companies such as Boeing and Airbus. During the past years, this technique was developed in wind tunnels and some universities to perform noise source identification on scale airframes, main landing gear models, and aerodynamic profiles (used on airplanes, helicopter rotors and wind mills).

Introduction

Innovative ideas, methods, and technologies are needed for the design and development of low-noise aircraft. Improvements in physics-based noise prediction methods, high-resolution noise and flow measurement techniques, robust noise control and mitigation strategies, and novel low-noise aircraft concepts are necessary to enable anticipated growth in air traffic worldwide while complying with increasingly restrictive controls on community noise levels. As part of a multi-pronged strategy to tackle this challenge, novel ideas are needed in advanced prediction and measurement techniques, innovative noise-reduction methods, low-noise propulsion/airframe integration concepts, unconventional low-noise aircraft configurations, and low-noise operations/noise reduction procedures.

Microphones arrays

The main purpose of a phased array measurement is to create a “sound picture” showing the most relevant acoustic sources at each frequency of interest for a complex body. This technique consists of a number of microphones arranged in a known pattern. The resulting “instrument” is called a microphones phased array. The pattern of the phased array can vary from a simple rectangular grid to a random placement of microphones. In recent studies, the most common arrangement for these microphones is a spiral configuration. The array pattern and size define the spatial resolution, the “useful” frequency range and the signal to noise ratio (SNR) of the results.

In order to obtain the desired results, the raw data from the microphones must be processed using a beam forming algorithm. The beam forming process assumes monopole sources located at every point in the desired scanning grid.

Using the phase delay from the monopole model, the microphone signals are added to determine the sound pressure level (SPL) at every point of the scanning grid. If the array is focused at an actual noise source, the microphone signals add constructively resulting into a large beamforming output. If a source is not present at that point in space, the signals add destructively yielding a low beam forming output. The same procedure is performed for each frequency of interest.

Beam forming is the discipline that takes a set of microphones, usually in an array, and a set of point source signals and tries to focus on one signal source to the exclusion of both noise and other signal sources. Delay and sum beam forming is quite true to its name as it merely takes the set of signals, delays and maybe weights them by varying amounts, and then adds them all together. The size of the delays is determined by the direction or point at which the set of microphones is aimed. Despite its simplicity, delay and sum manages to achieve optimal noise suppression for the case of a point source in a background of white noise. Of course, normal signal processing applies, and one can do better than just delay and sum if information about the signal other than location is known a priori. For example, if it is known that the signal is band limited and baseband, then a suitable low pass filter can be applied to further suppress noise.

Near field calculations are both more computationally intensive and accurate. If it is assumed that the microphones have some sort of center for distance, then the center can be designated as the origin for the coordinate system. A point source at a point (xs, ys, and zs) would then emit a signal s (t). A microphone at a point (xm, ym, zm) would then receive a signal m (t). Assuming that the signal propagates uniformly with speed v and that signal strength is equal to the original signal strength divided by the square of the distance, we can calculate the received signals/the received signals can be calculated.

If it is assumed that the array always operates in far field, with the source at a great distance from the sensors, an estimation in which the emitted spherical waves can be approximated by plane waves can be used. It is accurate in the limit where the distance between the microphones and the source is large enough so that the angle between the source and each microphone does not change significantly.

The beamforming process can be performed either in the time or the frequency domain. In case that frequency domain approach is used, the time domain signals are converted to the frequency domain by performing a Fast Fourier Transform (FFT).

With these data, a cross-spectral matrix (CSM) is generated for each frequency spectral line of interest. In order to “steer” the phased array to a particular point in the grid, the so called steering vector is used. This vector includes the theoretical phase delay for each microphone with respect to the scanning point as well as corrections in amplitude due to distance to the point. In some cases, corrections for microphone response are used to obtain the actual array output in the beamforming equation.

After the beamforming process is completed, the maximum lobe (main lobe) indicates the actual position of the sources at each scanning frequency. The signal to noise ratio (SNR) is determined by the difference between the main lobe and the side lobes, i.e. lobes not associated with a source but resulting from the spatial aliasing effects due the discrete nature of the array. Usually, the 3dB down zone from the maximum SPL at each frequency is assumed to be the size of the noise source. As mentioned before, the size of the “spot” or spatial resolution depends on the array characteristics. If more than one noise source is present at the same scanning frequency, they can also be “identified”. However, precaution must be taken to avoid confusing noise sources with lobes associated to louder noise sources.

Knowing the theoretical and/or experimental SNR of the array and the behavior of the side lobes is helpful during tests.

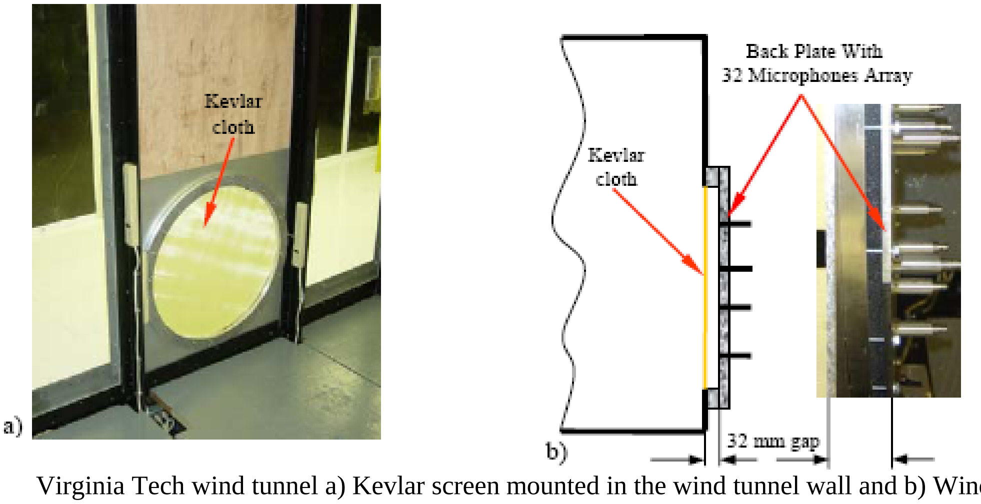
Problems regarding the beamforming techniques

During the beamformig process several complications may appear. Since all of the processing is done in a digital environment, all work must be done with samples of the signals and not with the signals themselves. In some cases, because of this, it is not possible to implement an arbitrary time shift as any shift must be done in increments of the sample period. To remedy this, the signals are interpolated digitally by up sampling them and then putting them through a low pass filter with cutoff corresponding to the amount of up sampling. An equiripple filter is chosen for the low pass filter as there appears to be no constraints as to the exact shape of the filter and because an equiripple filter would avoid the Gibb's phenomena found in a direct approximation of an ideal low pass filter. Using this interpolation, a greater resolution can be achieved in the time shifts, though the drawback is the large amount of additional data that must now also be processed. In fact, even though the concept of delay and sum is incredibly simple, the amount of computation that must be done because of the up sampling is often prohibitively high. It is impossible for the amount of interpolation to be too high, but if it is too low, then it is entirely possible that the direction of the source be inaccurate or entirely wrong as the algorithm will be unable to shift the signals to where they match enough. At a given sampling frequency, there is always the ”normal” aliasing phenomenon associated to the Nyquist Theorem, restricting the fully reconstructable signals to those that are band limited to half of the sampling frequency. Something similar occurs with array spacing, and if proper care is not taken, aliasing may occur in spatial dimensions. Using the spatial analogue of the Nyquist Theorem, the minimum spacing between microphones must be at most half the wavelength corresponding to highest frequency present. Thus, to achieve any resolution at all for higher frequency signals, smaller arrays must be used; however, with a smaller array, the precision with which a direction can be determined is diminished. It appears that there is an uncertainty principal at odds with beamforming in its spatial dimensions. The main focus of the process is to locate a source using an array of microphones and then to focus the array in the direction of the source, thus obtaining a greater suppression of noise than would be possible to obtain using only one microphone. Since the direction of the source is unknown, a solution is to scan for the source by sweeping all possibilities. This is where the far field approximation significantly reduces computational complexity. Using near field, any algorithm would be forced to evaluate all possible combinations of three coordinates. With far field, there are only two angles to deal with as opposed to three coordinates so there is far less to compute.

Some aero acoustic tests of airfoils, aircrafts models and high lift devices and other parts, are performed in an open jet anechoic wind tunnel and some are made in closed jet wind tunnels. During tests scaled models and full models are used.The resulting data are compared to the corresponding free-field microphone data. Usually in open-jet wind tunnels is easy to do tests because there are not so many reflected waves (there are no walls near the models) and there is more space for sensors and equipments but they are costly.

Tests in closed-jet wind tunnels are not so expensive, but require more care in data interpretation. This is partial because the microphone arrays are positioned in the geometric near-field (we can not use the approximation in which the source is assumed to be far enough away that the spherical waves it emits can be approximated with plane waves), the blockage factor of the model in a closed section tunnel is different to both an equivalent open-jet wind tunnel and an aircraft in flight, but also microphones must be used to suppress wind tunnel background noise. The subsonic wind tunnel from INCAS is a closed-jet type. In this case the best way to make aero acoustic measurements is by implementing arrays of microphones in the wind tunnel walls.

Since these arrays are to be used for aero acoustic measurements in a hard wall wind tunnel, some extra considerations are necessary in hardware fabrication. For instance, to avoid the hydrodynamic noise induced by the turbulence in the boundary layer of the wind tunnel, the microphones are recessed behind a stretched Kevlar cloth. This mounting technique was developed and tested at NASA Ames. Since Kevlar is acoustically transparent, the pressure fluctuations on the microphones due to the boundary layer unsteadiness are significantly reduced while allowing the sound to propagate through the cloth. On the other hand, given the high tension applied while stretching the cloth, the Kevlar appears as a hard surface to the flow.



Virginia Tech wind tunnel a) Kevlar screen mounted in the wind tunnel wall and b) Wind tunnel cross-section showing a schematic of the mounting of the phased array.

During the past years the development of quieter turbofan engines accentuated the airframe noise, thus becoming an important noise source in commercial aircrafts. While at take-off engines still are the dominant noise source, the airframe noise is as important as the engine noise on approach since engines are operating at low thrust. The main problem is that when approaching the airport for landing, the aircrafts fly over populated areas at low altitude.

Therefore, the high levels of radiated noise have a significant impact on community noise. The main components of the airframe noise are the high lift devices and the landing gears. In order to satisfy noise regulations imposed by the aviation authorities and some airports, the noise levels needs to be further reduced in future years. An important task for this goal is to clearly identify the noise sources, quantify their contribution to the overall noise emission, and develop and test practical noise control devices. Thus, it is critically important to identify the gear components that are mostly responsible for the flyover noise emissions. In this way, the design of noise control devices can be focused on those identified gear components.

Conclusions

The impact on the environment is very important, thus representing the main objective and the direct requirement of ACARE (the reduction of aerodynamic noise by 20%). Especially from the point of view of a new testing technology, this impact can be evaluated from two different perspectives which emphasize the importance of the proposed research.

The technologies that are used for the laboratory analysis systems are cutting edge technologies, with low noise levels and without any emissions of polluting substances. From the perspective of its application to a new generation of aeronautical products, by means of reducing the testing time and increasing the safety in the carrying out of the tests, the research projects offer the chance of limiting the negative effects upon the working environment (restrictive access for prolonged periods of time, protection and security systems, sound insulation etc.)

On the whole, the new proposed technologies will lead to a significant improvement of the environment conditions.

The major focus is the development of improved prediction methods and technologies for lower noise, lower emissions, and higher performance aircraft. The new test methods and validated prediction tools can be used to improve system tradeoffs for advanced concepts capable of meeting/ able to meet longer-term noise, emissions, and performance targets. It is expected that foundational research performed will lead to improvements in multidisciplinary design, analysis, and optimization tools, new experimental methods that provide fundamental properties and establish validation data, noise prediction and reduction technologies for airframe and propulsion systems, emissions reduction technologies, alternative fuels, and particulate measurement methods, improved vehicle performance through design and development of lightweight, multifunctional and durable structural components, high-lift aerodynamics, and higher bypass ratio engines with efficient power plants.

Advanced measurement techniques and experimental methods are required that would directly support research leading to reduced noise, reduced emissions, and increased vehicle performance. Innovative new approaches to current measurement techniques used in the testing and simulation environment are required to enhance critical facilities and capabilities used for the validation of advanced concepts. Areas of interest include structures and materials characterization, intelligent engine design, enhanced aerodynamic testing, and improved flight research techniques.

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