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AEROACOUSTIC TECHNIQUES USED FOR NOISE SOURCE IDENTIFICATION ON COMPLEX BODIES

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Abstract: *Over the last decades several approaches have been undertaken to asses detailed noise source identification on complex bodies. Traditional testing methods can not be applied in some cases at all and in others there are several limitations. Optical systems require a line of sight that may not be available. The state-of-the-art technology purposed is to use a large number of microphones whose signals are acquired simultaneously. The implementation of this technology is limited due to the cost of the instruments and the data acquisition system required.. Even though in previous years, this technique was developed in wind tunnels to perform noise source identification on scale airframes, main landing gear models, and aerodynamic profiles, more research is needed.*
Key words: *Noise, source, identification, acoustic.*

1. INTRODUCTION

Noise source identification implies to obtain the position and frequency of the dominant noise sources accurately. Innovative ideas, methods, and technologies are needed for the design and development of low-noise aircraft. The 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 essential 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 desired 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. 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 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. The microphone signals are added to determine the sound pressure level (SPL) at every point of the scanning grid using the phase delay from the monopole model. When the array is focused at an actual noise source, the microphone signals add constructively resulting into a large beamforming output. However, 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.

A suitable low pass filter can be applied to further suppress the noise, if it is known that the signal is band limited and baseband. Near field calculations are both more computationally intensive and accurate. Then, the center can be designated as the origin for the coordinate system, when the microphones are assumed to have some sort of center for distance. The beamforming process can then 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). 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 obtained by the difference between the main lobe and the side lobes, i.e. lobes not associated with a source but as a result of spatial aliasing effects due the discrete nature of the array. 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, it can also be "identified". However, precaution must be taken to avoid conflicting noise sources with lobes associated to louder noise sources.

2. EXPERIMENTAL SETUP

Tests performed in closed-jet wind tunnels are reasonably priced and require more care in data interpretation. This wind tunnel is a closed-jet type. This is partially 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 varies to both an equivalent open-jet wind tunnel and an aircraft in flight and microphones (W.O) must be used to suppress wind tunnel background noise. Our main goal is to develop and update the data acquisition systems by enhancing the investigation techniques of INCAS subsonic wind tunnel. For aeroacoustic evaluation tests we have implemented a circular ring array with 72 microphones and one video camera.

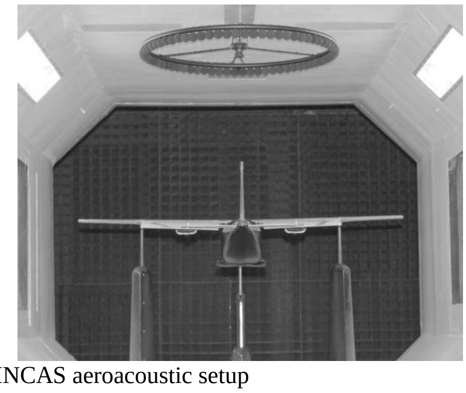

Fig. 1. INCAS aeroacoustic setup

In order to be able to verify acceptable flow quality, a calibration study was obtained prior. In this case the aeroacoustic studies were made on Aerotaxi model. Frequency dependent sensitivities of the individual microphones were taken from calibration sheets. No corrections were applied for microphone directivity. The ring array is large enough (one meter) to obtain high resolution at low frequencies and is placed on the ceiling of the wind tunnel. The distance from the center of the microphones array to the model is 0,9 m. Acoustic data from the array microphones were simultaneously read at a sample frequency of 96 kHz and a measurement time of 40 s. The test measurements were made using the microphones with and without windscreen for different model configurations (take-off and landing) and on different flow speeds. The signals from the microphones are A-weighted.

3. PROBLEMS REGARDING the BEAMFORMING TECHNIQUES

Several complications may appear using the phase delay from the monopole model. Even though all of the processing is done in a digital environment, all work is performed with signal samples and not with the signals themselves. In some cases, because of this, it is not possible to implement an arbitrary time shift since any shift must be done in sample period increments. To remedy this, the signals are interpolated digitally by up sampling then are passed through a low pass filter with cutoff corresponding to the amount of up sampling. Using interpolation, a greater resolution can be achieved in the time shifts. The drawback is the large amount of additional data that must now also be processed.

According to 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; eventhough 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. 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 others are made in closed jet wind tunnels. Tests are performed on scaled models and full models (W.O). Open-jet tunnel tests are simple because of minimal reflected waves (there are no walls near the models) and have extensive spaces for sensors and equipments. The drawback for this is the cost.

During my tests the signal from the microphones using windscreen has unwanted components on high frequency range but are present only on narrow frequency bands. The frequency of this signal peaks vary almost linearly with the wind speed. The peaks on high frequencies are substantially attenuated using the windscreen and low frequencies are increased on a wide band. The total sound pressure is then lower in the second case concluding that the windshield has a positive influence, thus allowing higher test speeds. If the signals follow to be post processed, the second case involves using a high-pass filter with low rejection slope. In comparison, the first case involves using a number of band-pass filters with improved rejection slope which use high processing power. The microphone performance can be improved by allowing even higher speeds on wind tunnel. Another effective way to make aero acoustic measurements is to implement microphone arrays in the wind tunnel walls or behind a protective wall.

4. CONCLUSION

As a result of these tests, the noise sources are idendified successfully. The dominant noise source was observed to be located at the trailing edge of the wings, in flaps and ailerons area.

Therefore, in the future the fluctuations produced by boundary leyer are minimized near micophones by recessing them behind a streched Kevlar cloth. This fabric is an acoustically transparent material which allows the pressure fluctuations on the microphones to be reduced due to the boundary layer unsteadiness, while propagating the sound. This mounting technique was already developed and tested at NASA Ames. On the other hand, by applying high tension, the Kevlar fabric acts like a hard surface wall

For the past years quieter turbofan engines development accentuated the airframe noise in turn becoming an important noise source in commercial aircrafts. While the take-off engines are still the dominant noise source, the airframe noise is as significant as the engine noise on approach when engines are operating at low thrust. The main components of the airframe noise are the high lift devices and the landing gears. As a result, the high levels of radiated noise have a significant impact on community noise. Thus, it is essential to identify the gear components that are mostly responsible for the flyover noise emissions and correlate the design of the noise control devices to those identified gear components.

The main focus still remains the development of improved prediction methods and technologies for lower noise, lower emissions, and higher performance aircraft. Advanced measurement techniques and experimental methods are then required. These would directly support research leading to reduced noise, reduced emissions, and increased vehicle performance. Innovative new approaches to current measurement techniques can be used in the testing and simulation environment to enhance critical facilities and capabilities used for the validation of advanced concepts.

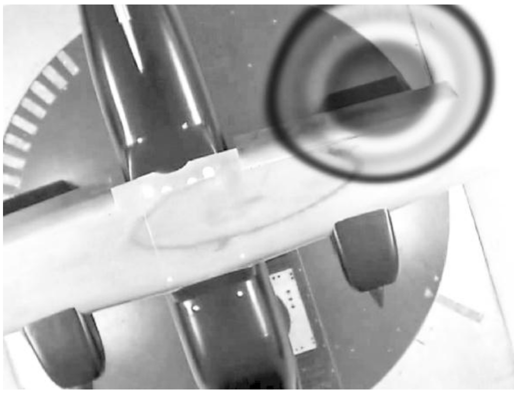

Fig. 2. Noise source identification

5. ACKNOWLEDGEMENTS

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