A STUDY TO INVESTIGATE WIND TUNNEL BACKGROUND NOISE FOR THE PURPOSE OF AEROACOUSTIC BEAMFORMING

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Abstract. *Even though aeroacoustic measurements have long become an important branch of research, additional advances have arisen from the use of microphone phased arrays and advanced beamforming techniques for the mapping of acoustic sources in closed-section wind tunnels. Designing and implementing a high-performance microphone phased array system require a feasibility study to evaluate how aeroacoustic measurements can be affected by the wind tunnel background noise level. However, most of the background noise spectral components are uncorrelated and can be eliminated from the cross-spectra matrix by means of an appropriated siganl processing procedure. This paper presents a study to better understand the features of the effective background noise measured by a 96-channel array installed in the University of São Paulo wind tunnel (LAE-1), aiming at quantify its level under several flow conditions and estimating a safely signal-to-noise ratio for aeroacoustic measurements. The results obtained by processing cross-spectral matrices, which were assembled from experimental data, provided an insight of how spatial and temporal averages affect background noise detection by a microphone array*.

Keywords*: Wind tunnel, background noise, Cross spectral matrix, Beamforming.*

1. INTRODUCTION

Frequency-domain beamforming methods based on microphone-phased array consist basically in convolving the array point spread function with the acoustic pressure field to isolate all coherent sound signals from the overall wind tunnel background noise. Beamforming is accomplished by algorithmically steering the microphone array towards each point of a scanning grid to estimate the mean squared value of the pressure field on the array in a post-processing step.

However, it has been verified that the beamforming algorithms performance can be highly sensitive to the signal-to-noise ratio, especially in a reverberant test facility, Mueller (2002). The signal is the genuine sound irradiated by self-noise mechanisms mechanically induced by incoming flow and the noise, in general, refers to the typical background-noise level. The main sources of background noise in a wind tunnel are the fan, wall boundary-layer turbulence, free stream turbulence, test-dependent hardware and strut self noise, including microphones (Mueller, 2002).

Measurements of the wind tunnel background-noise can be performed by using in-flow microphones equipped with a bullet-shaped forebody which replaces the standard grid for diaphragm protection. However, an in-flow microphone is under airstream conditions that are expected to be different from those found on the microphone array and the noise measurements do not take advantage of the spatial filtering process for incoherent noise achieved by correlating the signals from an ensemble of microphones.

Designing and implementing an aeroacoustic measurements system in a closed-section wind tunnel generally demand a wall-mounted microphone array. In this case, the microphone diaphragm is expected to be exposed to the noise generated by turbulent boundary-layer processes, even whether protected by a grid. Tensioned Kevlar cloth acting as an acoustic window has been proposed to reduce the influence of pressure fluctuations associated with boundary-layer turbulence on microphone phased array systems, Jaeger *et al*. (2000).

A turbulent boundary-layer flowing on the wall-mounted array develops vortex structures which act as coherent sound sources. The correlated pressure fluctuations associated with such vortex structures might encompass a number of microphones at the array, according to the free stream speed. In such cases, a higher noise level should be expected from cross-correlation involving microphone spacing smaller than those required for the vortices de-correlation.

A complete survey about the effect of the potential noise sources requires a carefully analysis to identify parts of the acoustic spectra which are contaminated by tonal or broadband noise, according to the nature of the contaminant noise. Estimating the signal-to-noise ratio on each contaminated portion of the acoustic spectrum is an important approach to evaluate the necessity of acoustic treatment for noise reduction. Acoustic treatment is specially needed if high noise level is present at frequency bands in which the genuine signal detection is expected to be found.

Array geometries are usually designed to maximize their wavenumber resolution and dynamic range response at a focal point and over specific frequency bands, Fonseca *et al*. (2010). The array pattern properties can be used to optimize near-field or far-field beamforming applications, Mueller (2002). However, the array optimization process by itself does not consider the influence of extraneous noise sources arising due to the reverberant nature of the test facility. Such spurious sources can be harmful to beamforming results if the aftermost signal processing is not able to suppress their effects.

This paper focuses on studying how design parameters, such as microphones spacing, affect the array's vulnerability for noise contamination. Standard approaches to noise level reduction based on signal processing procedures are applied to experimental data and discussed in some detail. Besides the spatial filtering effects obtained by correlating signals acquired at different microphone positions, noise reduction can be also achieved by averaging data blocks in Fourier space, Shin *et al*. (2007), and/or removing the cross-spectral matrix diagonal elements. After both spatial and temporal filtering processes based on cross-correlation approach, as well as diagonal removing have been accomplished, what remains in terms of spurious sound is the actual background noise to be taken into account for beamforming processing.

2. EXPERIMENTAL SETUP AND DATA ACQUISITION

The experimental data analyzed in this paper comes from the measurements performed at the empty chamber test of the LAE-1 wind tunnel. It is a closed-section subsonic wind tunnel located at the Aerodynamics Laboratory of the São Carlos Engineering School. The main features of this wind tunnel are: (1) cross section of the working section measuring 1.29 m in high and 1.67m in width and having 3m of useful length, (2) axial fan with eight blades propelled by a 150Hp AC electric motor which is controlled by a inverter via computer, (3) contraction ratio of 1:8, flow velocity ranging roughly from 10m/s to 45m/s and turbulence intensity around 0.25%. It was originally designed to host basic and applied aerodynamic experiments and recently has been acoustically adapted for aeroacoustic experiments too, Santana *et al*. (2010).

The data acquisition setup included a wall-mounted 96-channel microphone array, which was connected at a 106-channel National Instrument PXI-1042Q (8-slot 3U PXI Chassis, low 43dBA acoustic emissions). The slots of the PXI-1042Q were filled with the boards NI 4496 and NI 4498. Each PXI-1042Q modulo had sixteen 24-bit analog inputs and IEPE constant current signal conditioning. The modules operate with 114 dB of dynamic range and simultaneous sampling at rates up to 204.8 kS/s per channel. They are equipped with built-in anti-aliasing filters that automatically adjust the signal to the sample rate and also with software-selectable input gains up to 30 dB. The NI 4496 and NI 4498 modules use a method of A/D conversion known as delta-sigma modulation. With this technology, input signals are oversampled or sampled at many times according to the selected data rate and then subjected to a digital filter. The acquisition software, which is based on a National Instruments platform, has been specially developed by Embraer employees for aeroacoustic measurements in wind tunnel or fly-over conditions by using a large number of microphones.

The array was designed according to a logarithmic single-arm spiral pattern. However, the array configuration can largely vary if sub-arrays are considered. The array was assembled with ¼ inch microphones type 40PH manufactured by GRAS Company. They have a useful frequency range up to 20 kHz, dynamic range around 135dB and integrated CCP preamplifiers as well as built-in TED chips allowing that each microphone to be programmed as an individual unit. The 40PH microphone requires a constant-current power supply, or any other CCP compatible power supply. Fig. 1 represents the microphone array layout that was based on array geometries found in (Mueller, 2002).

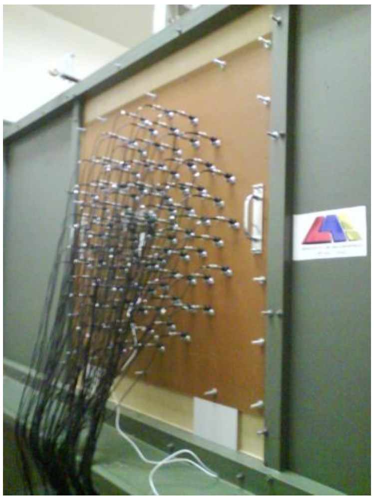

Figure 1. Microphone array configuration backside as used at the LAE-1 wind tunnel for aeroacoustic data acquisition. On the inside, the microphones were flush-mounted with an ambartex plate surface.

The microphones were equipped with standard grid protection and flush-mounted. This setup exposes the microphone grids to the hydrodynamic noise induced by boundary-layer unsteadiness occurring on the array. Approaches for preventing this occurrence consist in recessing the microphones or protect them behind a strongly stretched Kevlar cloth, Jaeger *et al*. (2000), Shin *et al*. (2007).

2. POST PROCESSING METHODOLOGY BASED ON CROSS-SPECTRAL MATRIZ

A cross-spectral analysis based on Fast Fourier Transform (FFT) was employed here to analyze the experimental noise data measured at the empty test chamber of the LAE-1 wind tunnel by a logarithmic pattern 96-element array. The raw input signals for post-processing were time-series collected under some wind-tunnel operation conditions. A smoothing procedure based on averaging process was used to reduce the random errors from the raw data (Bendat and Piersol, 2010).

Equation (1) represents a standard approach for the spectral estimation, $X[k\Delta f]$, of a discrete-time signal, $x[n\Delta t]$, by using direct Fourier Transform. In practice, such a signal is a digitized sequence of root mean square value of a pressure signal measured by any microphone in the array.

$$X[k\Delta f] = \frac{1}{N}\sum_{n=0}^{N-1} x[n\Delta t]w[n\Delta t]e^{-i2\pi kn/N}, k = 1,\ldots,n,\ldots,N \tag{1}$$

In Eq. (1), $\omega k = 2\pi k/N$ is the circular frequency, which is the multiple of the fundamental frequency $\omega_0 = 2\pi/N$, in units of radians per second. Because the angular frequency increases by integer steps, ω_0 is also the angular frequency resolution, defined as the spacing between two adjacent frequencies in the spectrum. Now, suppose a discrete signal has been divided into L data blocks of length N and windowed by a weighting function $w(n\Delta t)$ for spectral estimation on a digital computer. The parameter of interest here is the physical frequency resolution defined as $\Delta f = 1/N\Delta t$, where Δt is the sampling time and $N\Delta t$ is the duration of the discrete-time N-sized signal. However, if the discrete signal $x[n]$ is sampled at the frequency f_s, the frequency resolution can be redefined as $\Delta f = f_s /N$.

For spectral analysis, the sampling frequency f_s is commonly chosen to prevent aliasing in a frequency bandwidth of interest. For a given f_s, the frequency resolution will be adjusted in terms of the number of samples N in the data block. Index $K\Delta f$ designates the kth spectral component, satisfying the Nyquist cutoff frequency constraint, $K\Delta f \leq f_s / 2$.

Cross-correlation, or cross-spectral matrix (CSM), is assembled by performing a cross-spectrum operation for each pair of microphones and arranging them in an upper or lower triangular form. Because it is a Hermitian matrix, denoted by symbol (*), the complementary triangular elements can be achieved by a simple complex conjugate transposition. The elements of the main diagonal are designated the microphones auto-spectra.

Equation (2) represents the cross-spectrum G_{xy} obtained by cross-correlating spectra X and Y - each one associated with an arbitrary output channel - which are evaluated according to the Eq. (1). As each microphone is laid on a specific position at the array, it is easy to see that this operation provides spatial information for noise detection. W is a scale factor associated with the type of windowing function applied. Averaging is taken over all the L data blocks to reduce spectral variance and uncorrelated noise level.

$$G_{xy}[k\Delta f] = \frac{1}{LW}\sum_{l=1}^{L} X_l^*[k\Delta f]Y_l[k\Delta f] \tag{2}$$

By means of Eq. (2), one 2-D cross-spectral matrix is assembled after evaluating $(m!)/[2!(m-2)!]$ cross-spectra for each frequency component, in which m denotes the number of microphones. A 3-D matrix is generated by stacking 2-D matrices in a number of steps which match number k of the spectral frequency component defined according to the frequency resolution. In the spectral analysis conducted here, a 96-element array generates 4560 cross-spectra for each of the 255 frequency components.

The evaluation of a 2-D cross-spectra matrix provides physical and statistical contents about the sound field reaching the array (noise plus genuine signal), at a specific spectral frequency. On the other hand, selecting a pair of microphones, which means selecting two pairs of spatial coordinates (x,y), and looking at the cross-spectral distribution along the frequency axis provide an insight of how the level of the measured sound field behaves according to the spectral frequency at a specific region of the array. Such an approach was employed here to study the influence of the microphone spacing on the cross-spectra levels by taking the average of the cross-spectra for each pair of microphones over a frequency-band of interest.

The reason for focusing on a cross-spectral matrix is that the mapping of acoustic sources by means of frequency-domain beamforming approach adopted here consists in evaluating the convolution integral of the array's point spread function with all the sources generating coherent sound fields. However, the available physical information about sources location and intensity can be only accessed from the cross-spectral matrix. Even if the background-noise level were enough low to affect any aeroacoustic measurement in a close-section wind tunnel, it would be still constrained in some level by the physical limitation of the array design. In general, the drawbacks for accurate aeroacoustic measurements come from the parasitic noise contaminating the genuine signals. When this is the case, the signal-to-noise ratio can be better investigated from the CSM, and proposals aiming at noise reduction via wind-tunnel acoustic treatment or a new array design can be conceived in a safe base.

3. RESULTS AND DISCUSSION

The first analysis consisted in studying how the microphone spacing affects the noise detection by the array of microphones. Such a study has potential to estimate the spatial extension of the coherent boundary-layer pressure fluctuations according to the free stream speed, Shin *et al.* (2007). The results of the spectral estimation have been normalized like pressure spectrum level, which is defined here as $10log_{10}\{(G_{xy}(f)\Delta f)/(20\mu Pa)2\}$, whether G_{xy} has been estimated as cross power spectral density.

In Fig. 2, the 4560 cross-spectra values corresponding to the 96-microphone array signals were evaluated and plotted as a function of the microphone spacing, in the Euclidean norm sense. The 60s long time record which originates the CSM by means of Fast Fourier Transform was sampled at 40960Hz. Digital signal processing employed block length of 512 points to provide spectral frequency resolution of 80Hz. Because the following result can be obtained at each spectral frequency component, it was taken an average over the frequency range from 80Hz to 20400Hz. Running averages over this long spectral range only provides an overall insight of how noise detection depends on the microphone spacing, without distinguish the spectral behavior of the noise at lower or higher frequencies. Free stream speed was 27m/s.

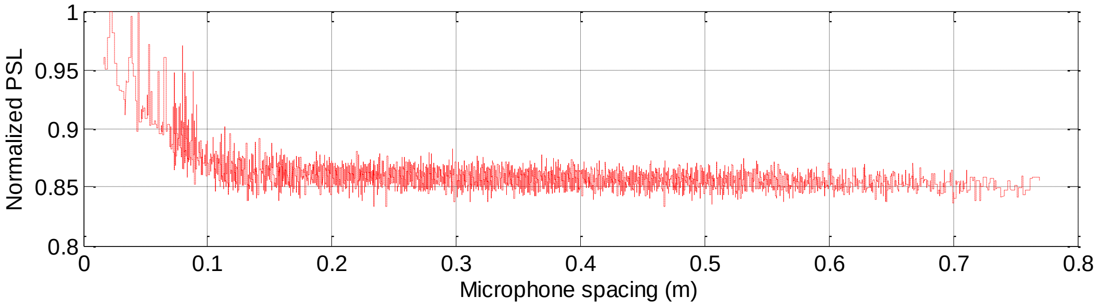

Figure 2. Normalized cross-spectral amplitudes according to the microphone spacing. Each point represents an average on the frequency range from 80Hz to 20400Hz, with frequency resolution of 80Hz, at 27m/s speed.

Figure 2 shows that the highest cross-spectra levels occur for microphones up to 0.1m apart. It implies that coherent pressure fluctuations associated with turbulent boundary-layer process can be only detected by closer microphones, which in general are located at the center of the array. Therefore, this can be a critical region to raise the estimation of the background-noise level. However, this analysis does not provide information of how the cross-spectra level depends on the frequency in which coherent pressure fluctuations develop.

This issue was addressed by taking the ratio between the mean cross-spectrum level derived from the channels pairs whose microphones do not exceed 0.1m apart and the mean cross-spectrum evaluated from the whole ensemble of channel pairs. Performing this procedure for each one of the 255 spectral components enables us to estimate in which frequencies, or region of the spectra, the noise level associated with coherent pressure fluctuations gets higher on the closer microphones. This analysis was carried out at 27m/s, 23m/s and 19m/s free stream speeds.

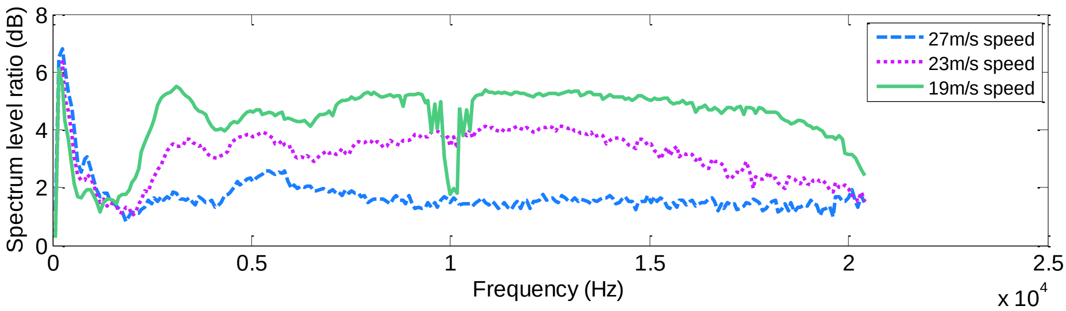

Figure 3. Mean cross-spectra ratio amplitude. The procedure was carried out for each spectral component at 19m/s, 23m/s and 27/s speeds.

In Fig. 3 it is possible to indentify pronounced peaks around 160Hz for 19m/s and 240Hz for both 23m/s and 27m/s speeds, indicating in which frequencies the pressure surfaces in the boundary-layer maintain stronger spatial correlation according to the flow speed. At these frequencies, the noise levels detected by the selected group of closer microphone are 6dB higher than the noise detected by the array as a whole, on average. It is also evident that the importance of the boundary-layer coherent noise for the total background noise level increases as the flow speed decreases. Increasing the air speed probably rises the level of the coherent noise originated by the free stream passing through the test chamber, or even in other part of the wind-tunnel circuit, which is detected by the array over the most part of the frequency range. Analyses in particular regions of the frequency range are cases apart.

Next, a study was conducted to better understand the spatial distribution of the coherent turbulence-generated noise on the array in terms of the streamwise (x) and spanwise (y) directions. Let us set Δx and Δy as the distances between any pair of microphones along the x and y directions, respectively. The 4560 cross-spectra values were ranked to form two groups: one as a function of the increasing values of Δx and another as a function of the increasing values of Δy.

Figure 4 represents the cross-spectral mean amplitude values, which were averaged over the whole frequency range, as a function of the Δx and Δy parameters. It is possible to verify that high cross-spectra levels, or peak values, concentrate on shorter microphone spacings along the spanwise direction but distribute themselves for longer extention along the streamwise direction.

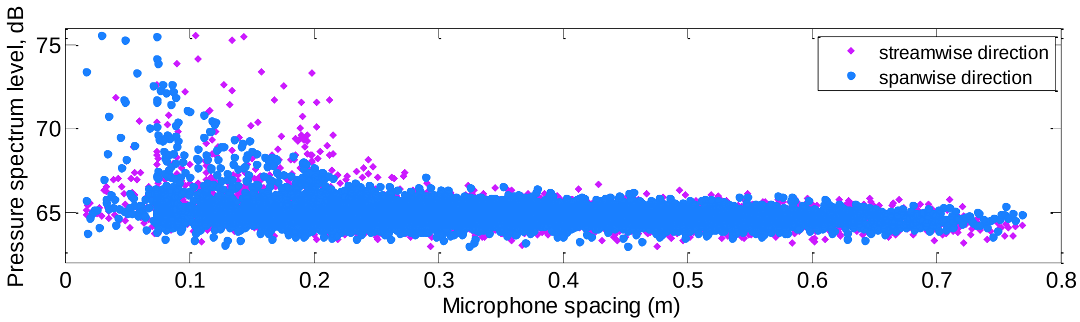

Figure 4. Spatial cross-spectral amplitudes distribution according to the Δx and Δy microphone spacing. Each point represents an average on the frequency range from 80Hz to 20400Hz . Free stream speed at 27m/s.

The result suggests that vortex-shaped structures generating noise remain longer correlated in the streamwise direction, which means that coherent noise is detected for a greater number of microphones when they are aligned along the x direction. If the signal processing provides a reliable spectral estimation, the extent of the region on which the cross-spectrum peak values concentrate, along the y-axis, gives us an idea of the size of the turbulent vortices irradiating coherent noise. Along the x-axis, the longest noise spatial correlation detected by pair of microphones as more scattered peak values is believed to be an effect of propagation of the coherent pressure fluctuation caused by vortices convection in the streamwise direction. However, this is a qualitative result and quantitative achievements require more detailed studies.

Spectral analysis is a powerful approach widely used to identify signals buried in either tonal or broadband noise. By using spectral analysis it is possible to focus on selected frequency bands to better estimate the signal-to-noise ratio.

The approach here consists in dividing the 96-microphone array into a number of 12 sub-arrays. The first sub-array was composed of the eight first microphones making up the logarithmic spiral layout and each ensuing sub-array was achieved by selecting and adding eight microphones at a time in the previous sub-array, always respecting the logarithmic spiral path. For each sub-array, the mean cross-spctrum level at each spectral frequency was evaluated as a cummulative process over the number of microphones. Fig. 5 to Fig. 7 show the cross-spectra levels for sub-arrays composed of 8 and 96 microphones when the wind tunnel operates at 27m/s, 19m/s and 11m/s, respectively.

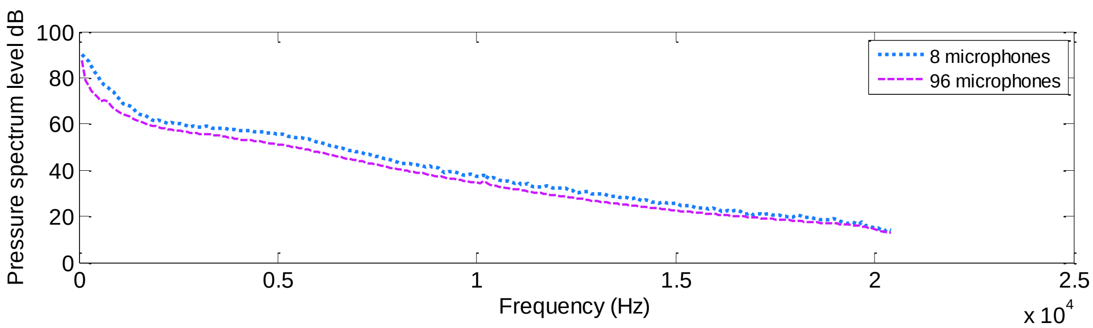

Figure 5. Mean cross-spectra amplitudes for sub-arrays composed of 8 and 96 microphones, at 27m/s speed.

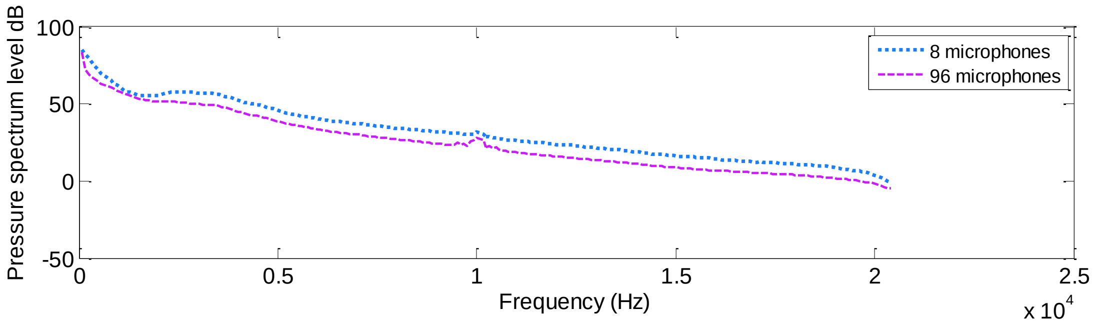

Figure 6. Mean cross-spectra amplitudes for sub-arrays composed of 8 and 96 microphones, 19m/s speed.

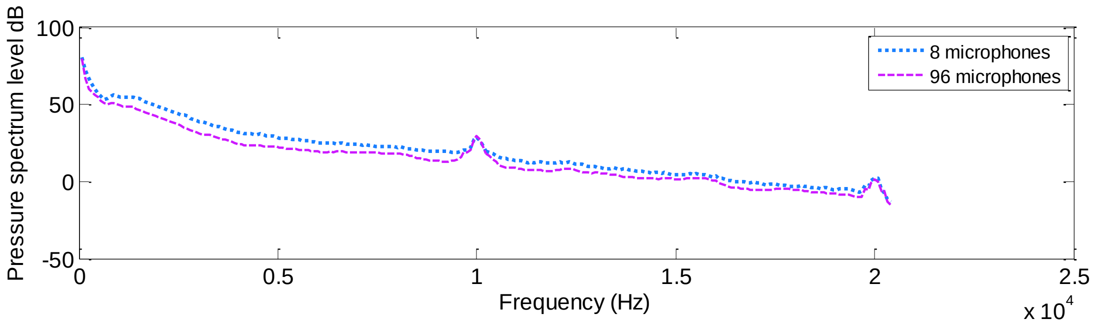

Figure 7. Mean cross-spectra amplitudes for sub-arrays composed of 8 and 96 microphones, at 11m/s speed.

From these results it is possible to confront cross-spectra levels for differents free stream speeds throughout the frequency range. At 27m/s and 19m/s does not exist significant like-tonal noise detected by the microphone arrays, or it is hidden under a higher broadband noise level. Therefore, broadband noise due to both free stream and turbulent boundary layer process, and probably fan noise at low frequency, will dominate the spectra from 19m/s speed in the empty wind tunnel. The spectral analysis reveal a more pronounced like-tonal noise centered at 10.000Hz for 11m/s speed. It can be an important point to be investigated, because noise peaks arising at the same frequency band of signals of interest, even whether covered by a higher broadband noise level, can be harmfull for beamforming accuracy.

Another analysis point focuses on how the number of microphones at the array acts to reduce the mean noise level by means of spatial filtering effect. In Fig. 8, spectral averages over the whole frequency range are presented as a function of the number of microphones making up each multiple of eight elements sub-array, starting at eight and ending in 96 microphones. 27m/s, 19m/s and 11m/s speeds values were considered.

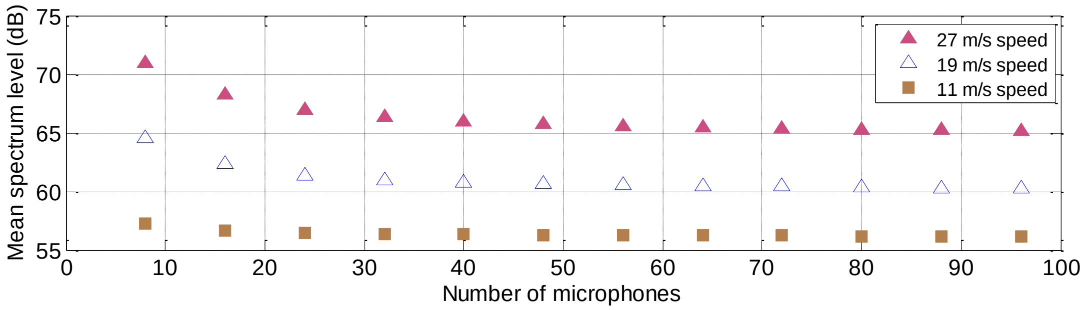

Figure 8. Mean cross-spectra amplitudes for sub-arrays composed of a multiple of eight microphones, at 27m/s, 19m/s and 11m/s speeds.

The above result provides an overall insight of how the mean cross-spectra levels come down as the number of microphones increases. By taking the noise level estimated from the sub-array composed of the first eigth microphones as reference, around 6dB of reduction was achived by cross-correlating the signals of the 96-microphone array at 27m/s speed. Furthermore, no significant noise level reduction was obtained by using more than 40 microphones. At 11m/s speed, no significant noise level reduction was achieved regardless of the number of microphones used.

The more pronounced noise level decay observed from sub-array composed of up to 24 or 32 microphones at 27m/s speed might be related with the fact of the coherent boundary-layer noise mainly concentrates on the closer microphones at low frequencies. As more microphones are included in the cross-correlation matriz, the noise induced by the free-stream turbulence prevails and makes the noise level redution reaches a plateau. Also, results not here included showed a little different noise level reduction trends as a function of the flow speed as the spectral analysis had been concentrated only at lower or higher frequencies.

This analysis indicates that it is difficult to set an empirical rule to quantify how much of noise level reduction can be achieved as function of number of microphones at the array according to the free stream speed for frequencies of interest. Thus, a basic signal processing procedure based on experimental data should be executed to find the real array capability to reduce the background noise level according to the number of instaled channels.

An important fact is that the background-noise level is not going to be further reduced by action of the beamforming techniques. Conventional beamforming algorithms add up constructively sound waves coming from a point grid basically by means of a delay-and-sum process. It increases the signal-to-noise ratio by reinforcing the coherent signal from a source. However, the noise level was already set during the acquisition process.

Up to now, the reuslts have focused on spatial averaging process to estimate the effect of the array design for background-noise detection. However, averaging process applied to data blocks during FFT calculation involving all pairs of channels has been taken into account in an implicit way. It is achieved by initially breaking a single time-record data segment of arbitrary length in an appropriated number of slices, each one having its length given by an integer number which is power of two. Generally, it is necessary truncate the original time series or zero padding each time slice in order to achieve the required number of data points per slice. After, a selected window weighting function is applied to each segmented-time slice (data blocks) before to perform the FFT operation on each one to form sequences of modified periodograms. Evaluating spectral quantities for each segment of samples and taking the average over the ensemble of blocks provides an approximated expected-value of the random signal which is necessary for estimating the cross-correlation between each pair of channels.

Background noise data were gathered by the microfone array in the empty test chamber during 60s. For each CSM evaluation, all channel data were segmented to form time-series of 30, 15, and 3.75s and so on, up to the minimum time-serie which roughly corresponds to one block size of 512 points. The number of averages associated with each FFT block size is given by $2*(T*f_s)/L$, with $T=N\Delta t$. The factor 2 is due to the 50% overlapping factor. However, the number of averages here ranges from 9600 for 60s acquisition time to roughly zero when the time-series length approachs to the FFT block size.

Figure 9 outlines the effect of this averaging procedure for filtering random noise. The lowest bounding line represents the mean amplitude of the cross-spectrum that was obtained by averaging 9600 data blocks. The higher bounding line corresponds to the result obtained from the minimum number of averages. The empty wind tunnel was run at 27m/s.

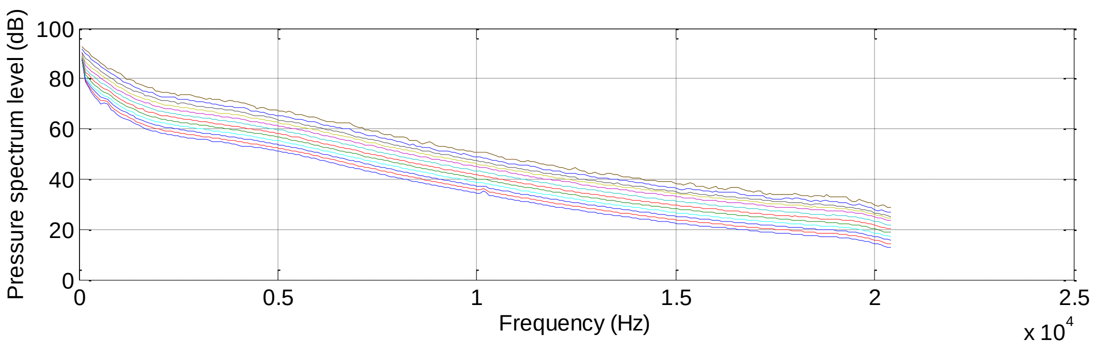

Figure 9. Spectra level mean values were estimated by averaging different number of data blocks. The block size is not changed and the number of blocks was set by truncating the time-signal before to perform the FFT.

To complete this approach, Fig. 10 shows the mean cross-spectra levels decay tendency as the number of averages increases. The gap between each adjacent line quantifies the noise level reduction achieved by doubling the number of block data to be averaged. Thus, the gap between the lowest and the highest lines indicates how much noise level reduction was achieved by performing 9600 averages.

On average, 1.5dB of noise level reduction is expected by doubling the number of data block to be averaged. However, such level of noise reduction can be dependent on the frequency range in analysis or modulation phenomenon associated with microphone spacings, Shin *et al*. (2007).

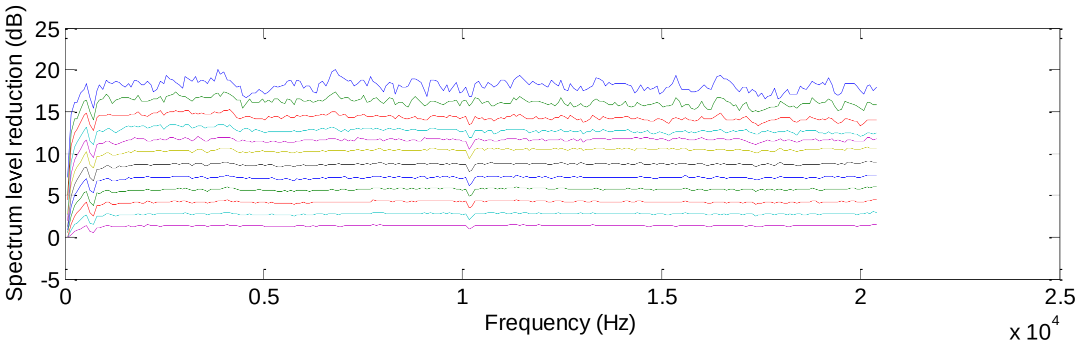

Figure 10. Spectra levels mean values differences according to the number of averages employed in FFT calculations. For unchanged block size, the number of averages doubles as the size of the time-series doubles.

The overall variation of the spectra levels show that the effect of taking 9600 averages over data blocks was a reduction of around 16dB in the background noise level measured at 27m/s. Technically, the lowest noise level to be achieved here by means of averaging process lies on 135dB, which is the dynamic range of the acquisiton system, below the auto-spectrum level, which is the highest noise level being measured at a moment.

Approximately 1,5dB of noise level reduction has been achieved by doubling the number of data block to be averaged. It is observed throughout the frequency range, especially when a larger number of averages are executed.

Next, the effect of the number of averages on the background noise level reduction is accounted for in terms of the acquisition time, becouse in general it is an empirically adjustable parameter in experimental procedures. In Fig. 11 and Fig. 12, there are thirteen points, each one representing the mean amplitue of one cross-spectrum plotted in Fig. 9.

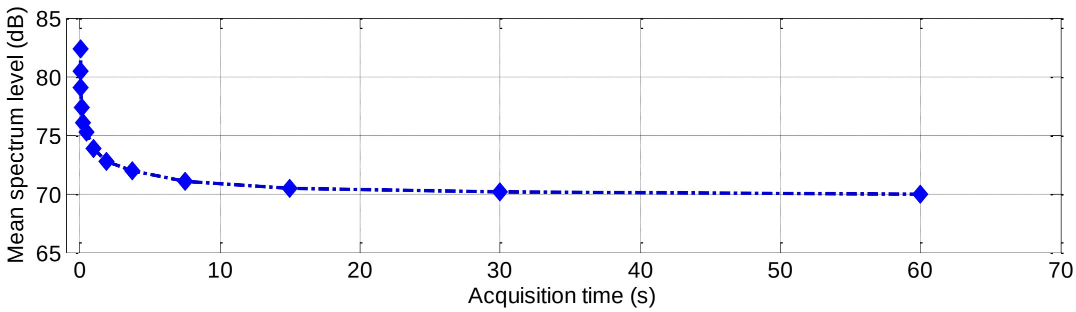

Figure. 11. Mean cross-spectra levels reduction as a function of the acquisition time. The averaging process over cross-spectra values were accomplished at the frequency band from 80Hz to 6800Hz.

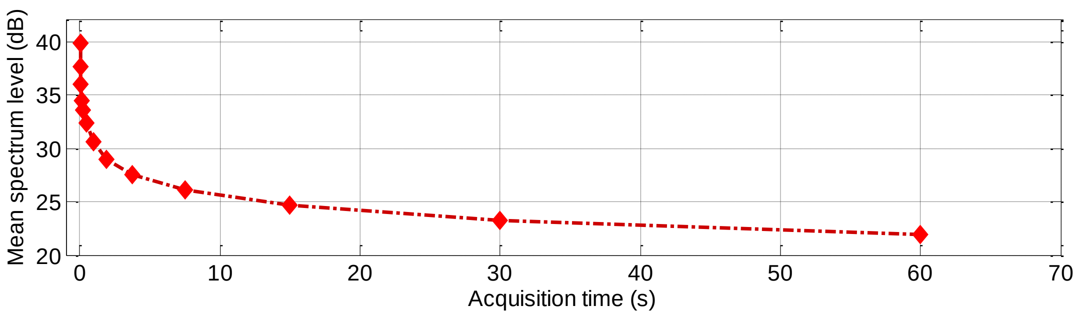

Figure. 12. Mean cross-spectra levels reduction as a function of the acquisition time. The averaging process over cross-spectra values were accomplished at the frequency band from 13600Hz to 20400Hz.

These results ilustrate the fact of the random noise level go down around 1.5dB by doubling the number of block data to be averaged, in terms of the acquisition time. At the beginning, the 1,5dB gain is achieved just by doubling a very short time-series, but as the time-series increase, achieving the same gain requires longer FFT computations.

According to Fig. 11, by applying 1600 averages over a signal of 10s long provided a noise level reduction around 10dB. On the other hand, Fig. 12 shows a noise level reduction around 15dB by applying 1600 averages over a signal of 10s and 18dB when a 60s long signal was averaged 9600 times.

Diagonal removal beamformig, which consists in removing the CSM main diagonal elements, is an approach commonly used to reduce the effect of incoherent background noise for signal-to-noise estimation, including the self-noise associated with flush-mounted microphones. It might be a suitable method to improve the signal-to-noise ratio when microphones are not recessed or protected by an acoustic window, such as Kevlar cloth.

The main diagonal elements from the CSM are associated with the auto-spectra of the microphones that provides an estimate of the absolute pressure field level impinging on the array at each frequency band. In this case, the background noise level reduction due to the spacial cross-correlation filter effect is not present. A more complete study about diagonal removal for beamforming purposes can be found in (Ravetta, 2005).

Aiming at quantifying the effect of removing the CSM diagonal elements, the mean auto-spectra were compared to the cross-spectra evaluated with and without removing the diagonal elements. A tweeter was used as acoustic source to generate a tonal signal centered around 10kHz, when de empty wind tunnel was run at 27m/s. The signal was canalized to prevent signal distortion due to the contact between the tweeter and the air flow. The results are shown in Fig. 13.

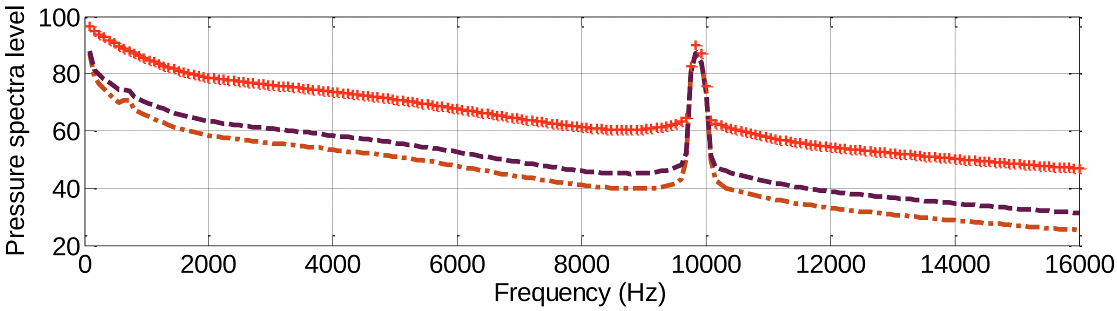

Figure 13. Spectral estimation for: a) auto-spectrum (+); b) cross-spectrum without elements of the main diagonal to be removed (−−) and c); cross-spectrum with elements of the main diagonal removed (− .). A like-tonal singal was added to the background noise measured at 27m/s.

Figure 13 shows that around 5dB of cross-spectrum level reduction was achieved, on average, when the elements of the main diagonal were removed. It is a real gain in the signal-to-noise ratio that is achieved by purely zeroing the CSM diagonal elements before the beamforming processing. Also, a gap of around 20dB between auto-spectrum and diagnonal-removed cross-spectrum mean levels represents the spatial filtering effect over incoherent noise that is achieved by simply cross-correlating the microphone signals. Unlike the broadband noise, the amplitude of the tonal signal was practically preserved. Such a fact ensures the feasibility of using such techniques to improve signal-to-noise ratio for the mapping of coherent sources.

Figure 14 ilustrates the effect of removing, or zeroing, the diagonal elements from the CSM. The 3-D graphics show a strong dominance of the auto-spectra against the cross-spectra level. Removing the CSM diagonal elements makes easier to map the cross-spectral topology. In this low-frequency case - 400Hz - it is possible to indentify higher cross-spectra levels associated with the first microphones, which are the same ones located at the center of the array. This is a graphical approach to better visualize the relationship between auto-spectra and cross-spectra levels at each frequency and also try to find possible regularities in the cross-spctral distribution according to the microphone array pattern.

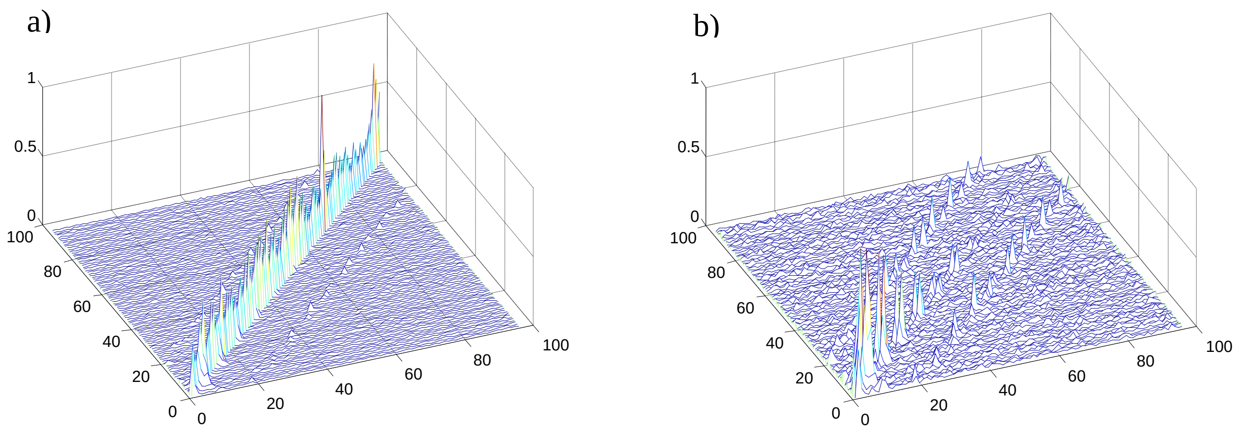

Figure 14. Normalized CSM: a) with oringinal diagonal elementes and b) with diagonal elements canceled. This analysis was performed at 400Hz and 27m/s speed. CSM was composed of 96 microphones.

4. CONCLUSIONS

An array composed of 96 flush-mounted microphones was employed here to investigate the main features of the LAE-1 wind-tunnel background noise at low Mach number. An approach based on CSM method was addressed to quantify the authentic noise for future signal-to-noise ratio assessment. At low frequencies, the background noise is probably more influenced by boundary-laryer pressure fluctuations on the flush-mounted microphones and noise generated by Fan.

It was also verified that stronger coherent pressure fluctuations can be detected by a group of closer microphones and the lower the free stream speed, the higher the influence of the coherent pressure fluctuations with respect to the mean noise level estimated by the microphone array as a whole. In general, the achievements provided an overall insight of how the background noise level depends on the airspeed and the frequency at which the typical flow-induced noisy processes develop.

An effective background-noise level reduction was achieved by averaging data blocks during FFT procedure or removing the elements from the CSM main diagonal. However, such approaches were verified to be effective to reduce the uncoherent broadband random noise, which is mostly associated with turbulent free stream effects.

Although studies aiming executing wind-tunnel acoustic treatments require more detailed spectal analyses to identify and eliminate spurious noise sources, the budget of studies here presented should be carried out as a previous study to better understand the noise features detected by the microphone array and eliminate steps to optimize the costs involved in the wind-tunnel acoustic treatment process.

5. ACKNOWLEDGEMENTS

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