PHASED ARRAY TECHNOLOGY DEVELOPMENT AT VIRGINIA TECH: APPLICATION TO LANDING GEAR NOISE SOURCE IDENTIFICATION

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Key words: aeroacoustics, noise source identification, phased array.

Abstract. *Phased array technologies have been developed at Virginia Tech to perform noise source identification of aerodynamic noise sources, i.e. noise due to flow. A 32 element phased array was designed, tested an implemented. Details on the design process are shown. Preliminary testing in the anechoic chamber and the wind tunnel was conducted to evaluate the performance of the array in out-of-flow and in-flow condition. Results for the most significant preliminary tests are also shown. As part of this work, noise source identification on a 26%-scale 777 main landing gear model was performed at the Virginia Tech 6- by 6-foot Stability Wind Tunnel. Illustrative results of the phased array performance are presented here.*

1 INTRODUCTION

Over the last few decades, many approaches have been undertaken in order to asses detailed noise source identification in complex bodies, i.e. aircrafts, cars, machinery. Noise source identification implies to accurately obtain the position and frequency of the dominant noise sources. The state-of-the-art technology for this purpose is the used of a large number of microphones whose signals are acquired simultaneously, i.e. microphone phased array[1-7]. 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 year, with the financial support of NASA Langley Research Center, this technique was developed at Virginia Tech to perform noise source identification on a 26%-scale Boeing 777 main landing gear model. As a result, a 32-element phased array was constructed and the computer codes to beamform and process the data were developed. This paper describes the development of this phased array technology and its application to noise source identification in landing gears.

2 PHASED ARRAY TECHNOLOGIES

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 defines 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 beamforming algorithm. The beamforming process assumes monopole sources located at every point in the desired scanning grid. The complex pressure of a monopole source is given as:

$$p = \frac{A}{r} e^{-i(\omega t - kr)} \qquad (1)$$

where: r is the distance from the observer to the source.
p is the acoustic pressure at a distance r from the source.
A is the amplitude of the noise source.
ω is the frequency of the source.
k is the wavenumber.

Using the phase delay from the monopole model, e^{-ikr}, 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 beamforming output. The same procedure is performed for each frequency of interest.

The beamforming process can be performed either in the time or the frequency domain. In this work, the frequency domain approach was used. For this purpose, the time domain signals are converted to the frequency domain by performing a Fast Fourier Transform (FFT). With this data, a cross-spectral matrix (CSM) is generated for each frequency spectral line of interest. The CSMs contain the cross-spectra between all microphones in the off-diagonal elements and the auto-spectra of the microphones in the diagonal. 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, as shown by:

$$\hat{e}_n = e^{-ikr_n} \qquad (2)$$

where: $\hat{e}_n$ is the n-th component of the steering vector.

r_n is the distance from a point in the grid to microphone n.

The array output is then obtained using the beamforming algorithm as follows

$$p = \hat{e}^T \ G \ \hat{e} \qquad (3)$$

where: p is the pressure at the scanned point.
$\hat{e}^T$ is the transpose complex conjugate of $\hat{e}$
G is the cross spectral matrix

for each point of the grid and for each spectral line of interest. 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 (mainlobe) indicates the actual position of the sources at each scanning frequency as shown in figure 2. The signal to noise ratio (SNR) is determined by the difference between the mainlobe and the sidelobes, 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

sidelobes is helpful in this task.

3 PHASED ARRAY DESIGN

In this research, several factors were taken into account for the design of the array: the available number of microphones (max. 32), the upper frequency range of the microphones (~19 kHz), the frequency range of interest (4-18 kHz), the distance to and size of the source, and the desired signal to noise ratio (SNR). The simulation of several array sizes and patterns rendered the best results for a logarithmic single-arm spiral. The equation that gives the microphone coordinates for a spiral array can be found in the literature[8]. However, better results were obtained modifying the position of the microphones along the spiral, l_m, from:

$$l_m = \Delta l \left(m - 1 \right) \quad (4) \qquad \text{to} \qquad l_m = \Delta l \left(\sqrt{m} - 1 \right) \qquad (5)$$

where Δl is the delta arc-length between sensors and m=1,2,..32.

As a result, the Virginia Tech (VT) phased array is a logarithmic single arm spiral with 32 PCB microphones type 130B10. It has an outer diameter of 0.71 meters and a minimum SNR of 8 dB at 20 kHz. It means that the worst sidelobe is at least 8 dB below the level of the mainlobe at all frequencies. The VT array pattern is shown in Figure 1.
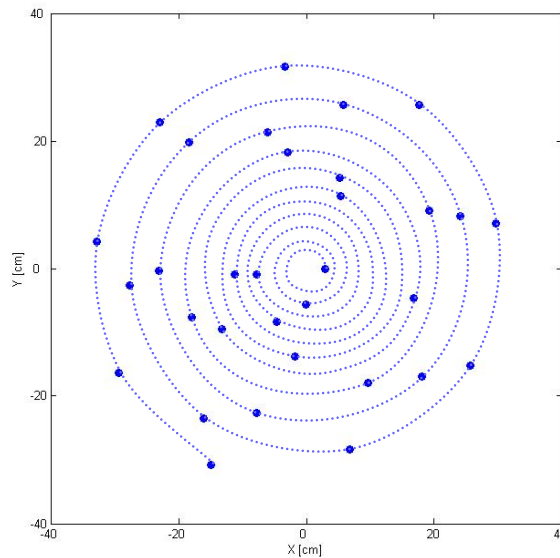

Figure 1: Single arm spiral phased array designed at Virginia Tech.

Figure 2 shows the beamforming simulation of a monopole source driven at 10 kHz positioned at the center of a plane parallel to the array at a distance of 0.91 meters. As can be seen, the SNR is at least 10 dB for the shown frequency. The maximum level was set to 0 dB

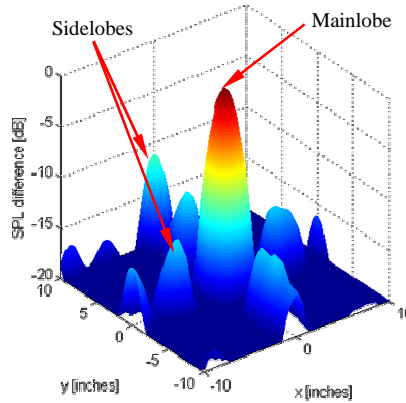in order to use relative levels. The lower level was set to -20 dB for ease of visualization.

Figure 2: Beamforming simulation for a monopole source at 10 kHz.

Since the beamforming algorithm makes use of the relative phase between the microphones and the microphone spatial locations a calibration must be performed to account for errors in the relative phase of the microphones and the positioning of the microphones. The calibration is performed in an anechoic chamber using a speaker driven with white noise. As a result of this test, a calibration matrix is generated for each frequency spectral line and then used as an input in the beamforming code. Basically, these matrices contain the phase delay for each microphone needed to maximize the array output at the known position of the speaker for every scanning frequency. These phase delays are then used to correct the data from an actual scanning of a complex source.

4   ARRAY HARDWARE AND MOUNTING

Since this array was to be used for aeroacoustic measurements in a hard wall wind tunnel, some extra considerations were necessary in the actual hardware fabrication. Thus, to avoid the hydrodynamic noise induced by the turbulence in the boundary layer of the wind tunnel, the microphones were recessed behind a stretched Kevlar cloth. This mounting technique was developed and tested at NASA Ames[9]. 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. A schematic and pictures of the setup are shown in Figure 3.
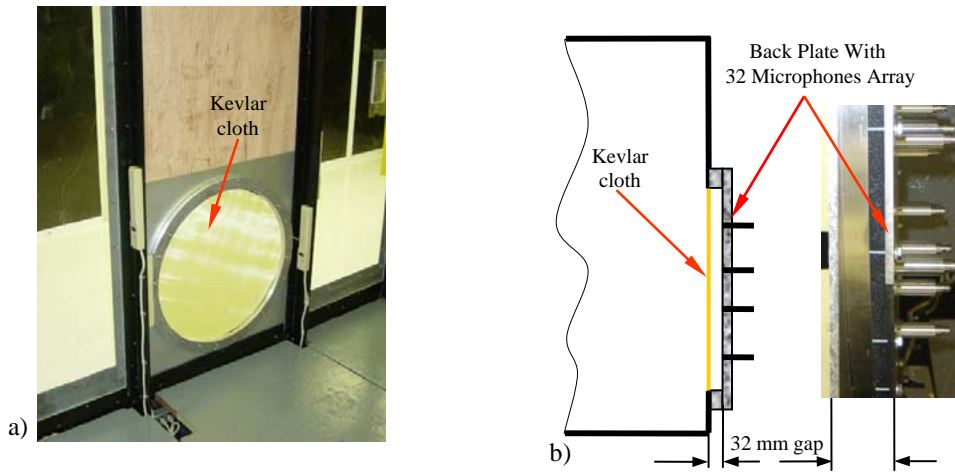
Figure 3: a) Kevlar screen mounted in the wind tunnel wall and b) Wind tunnel cross-section showing a schematic of the mounting of the phased array.

5 PRELIMINARY TESTING

To test the performance and capabilities of the array and the beamforming code, a variety of preliminary tests were performed. However, only the most significant ones will be described in this section. The first step was to test the array using a speaker, driven with a 10 kHz pure tone, in an arbitrary position inside an anechoic chamber. After beamforming, the position and frequency of the source were determined successfully. Figure 4 shows the setup for this test and some beamforming results for 1/12th octave frequency bands with center frequencies of about 10 and 18 kHz. Performing the analysis in 1/12th octave bands to reduce the number of beamforming maps to be analyzed is a common practice in this field. In Figure 4, the array output levels were cutoff below -20dB for ease of visualization.
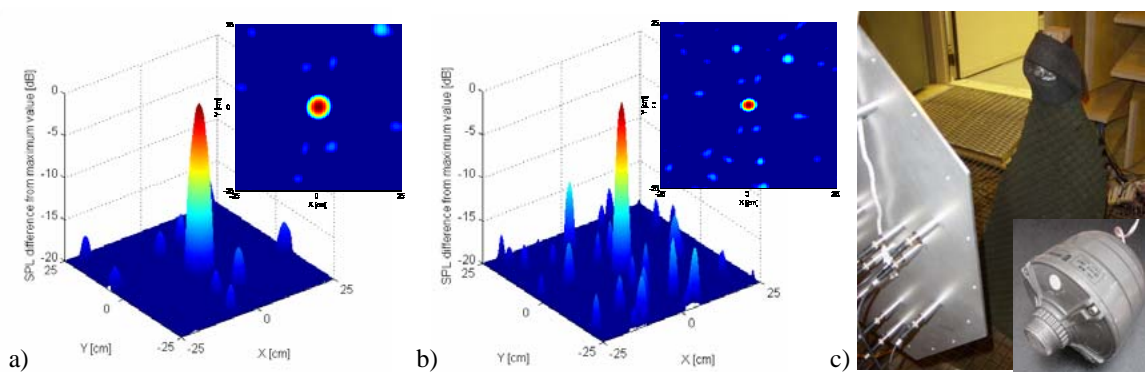

Figure 4: Anechoic chamber calibration results at: a) 10 kHz and b) 18 kHz. c) Phased array mounted in the anechoic chamber and detail of driver unit.

The purpose of the second preliminary test was to study the capabilities of the array when

operating in a reverberant environment. To this end, the speaker was driven again with a 10 kHz pure tone, in an arbitrary position inside a closed wall wind tunnel as shown in figure 5a. The beamforming results shown in figure 5b indicate the highest SPL for 10 kHz at the correct speaker location. In other frequencies, the mainlobe is at least 60 dB below the one found at the input frequency of 10 kHz. Therefore, the results were successful in terms of frequency and spatial location of the source. The spatial resolution was also in good agreement with the simulations.
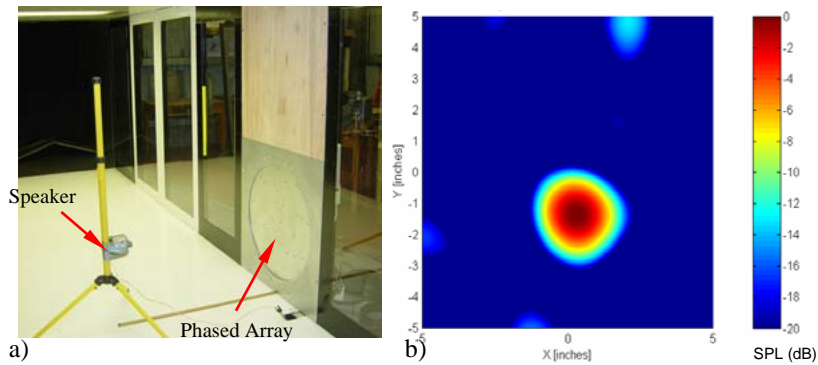

Figure 5: a) Test setup to evaluate no-flow performance of the array in reverberant environment, and b) beamforming results at 10 kHz showing the speaker position (right).

The third preliminary test was performed to analyze the array capabilities with noise sources in the presence of flow. For this purpose, a whistle was mounted on an airfoil in the wind tunnel test section. In this configuration, the flow in the tunnel drove the whistle producing a distinct tonal signal. Due to the compact size of the whistle, it behaves as a "monopole" source. Since the frequency of the whistle depends on the flow speed, this configuration was tested at several flow speeds Mach numbers. The setup and beamforming results for a flow speed of M=0.17 are shown in Figure 6. The whistle position was successfully determined and the maximum array level was found at the $1/12^{th}$ octave band #145 (center frequency approx. 4.34 kHz).
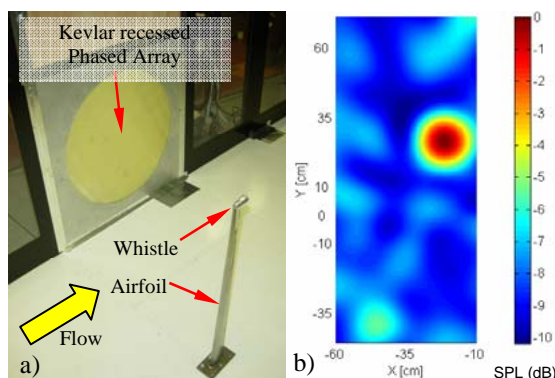

Figure 6: a) In-flow testing conditions using a whistle, and b) beamforming results at 4.34 kHz showing the whistle position.

The last preliminary test used the same whistle mounted on the door of a landing gear model to evaluate the performance of the system in the presence of a complex source. More details on the landing gear model will be given later. Figure 7 shows the setup and results. The maximum level for the whistle was again found for the $1/12^{th}$ octave band #145, as in the previous test, i.e. same flow speed. The whistle position was also determined successfully. The shifting in the whistle position due to the flow speed was the same in both tests and in agreement with the monopole equation with flow.
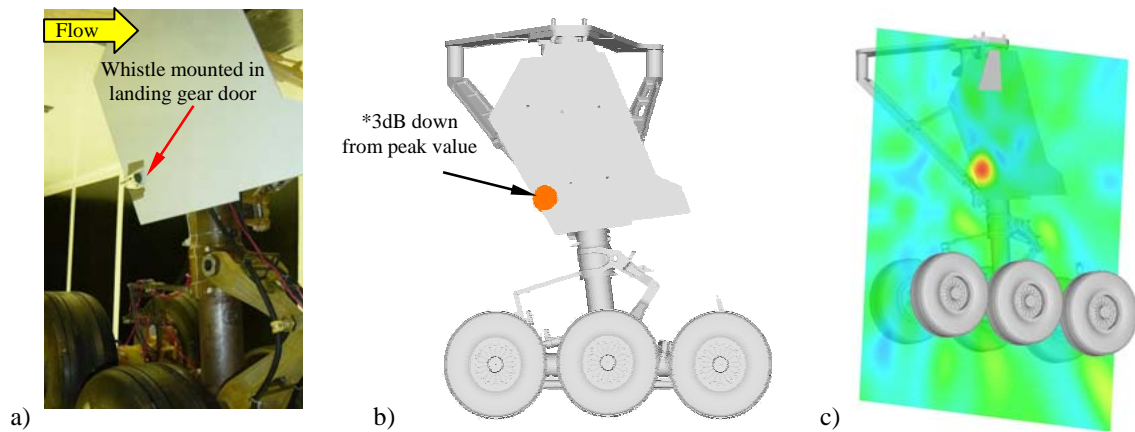

Figure 7: In-flow testing with complex source: a) whistle mounted in landing gear door, b) and c) beamforming at 4.34 kHz showing the whistle position.

6   APPLICATION TO LANDING GEAR NOISE

The main purpose of developing this technique was to apply it to landing gear noise. During the past years the development of quieter turbofan engines accentuated airframe noise, thus becoming an important noise source in commercial aircrafts[10]. While at take-off engines still are the dominant noise source, 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 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 to those identified gear components.

The high fidelity 26%-scale 777 main landing gear model used in this study was originally

tested under the AST/QAT (Advance Subsonic Transport/Quiet Aircraft Technology) program at the NASA Ames 7- by 10-ft wind tunnel[2]. The same gear model was later evaluated under the STAR (Subsonic Transport Aeroacoustic Research) program using a semi span model of the 777 at the NASA Ames 40- by 80-ft wind tunnel[1]. The model is shown in figure 8 in the actual landing position. However, for the tests performed at Virginia Tech, the model was mounted on the floor of the wind tunnel test section, i.e. upside down. This high fidelity model features all the major components and most of the details found in the original landing gear, including: oleo lines, cables, wheel hubs, brakes cylinders, hydraulic valves. The main structure of the model is made of steel and aluminum and the details are mostly made in stereo lithography up to an accuracy of 3mm in full-scale.

Figure 8: 26%-scale Boeing 777 main landing gear model in the Virginia Tech tunnel.

The experiments were conducted at the Virginia Tech Stability Wind Tunnel shown in figure 9. The facility is a continuous, closed jet, single return, subsonic wind tunnel with 24 foot long square test section of 6- by 6-ft. The tunnel is powered by a 600 hp DC motor driving a 4.26 m propeller providing a maximum speed of 300 km/h for the empty wind tunnel.
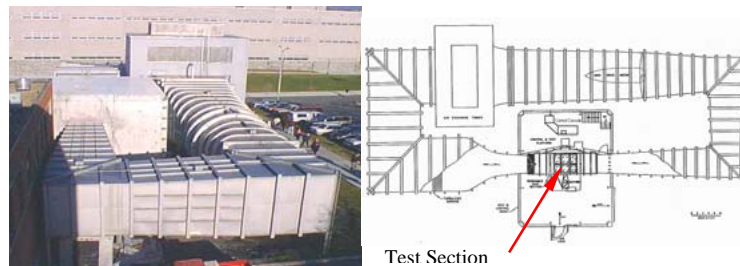

Figure 9: Virginia Tech Stability Wind Tunnel.

The noise source identification process for a couple of frequencies is shown in figure 10. In this case, the results for a plane parallel to the array are shown using a 2-D contour plot. As can be seen, the sources can be clearly identified from these maps in terms of position and

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frequency. Another way to look at the results is shown in figure 11. In this case a 3-D contour plot is shown where the lower cut-off value was set to 3dB below the maximum level for that frequency. This way of analyzing the data gives a more intuitive sight of the results and allows finding the correct position of the source in the axis perpendicular to the array. The detailed analysis of the noise sources on a 777 main landing gear can be found in the literature[11].
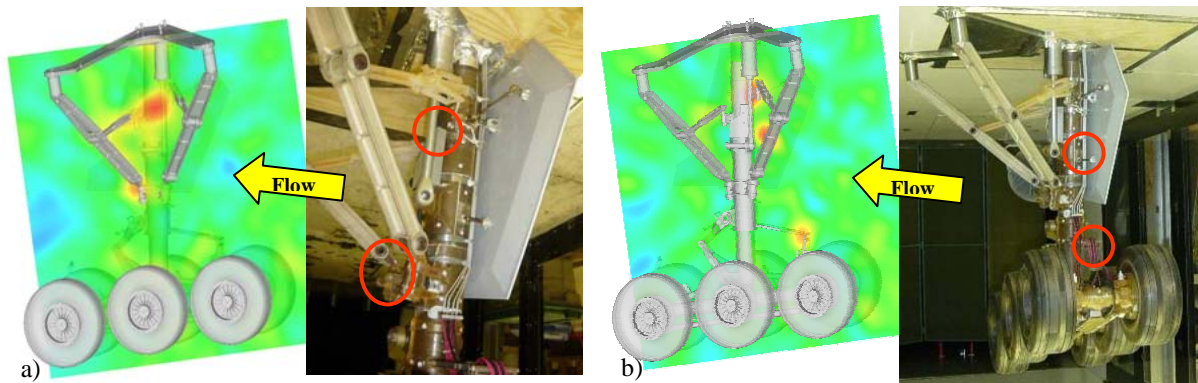

Figure 10: Noise source identification process at a) 5160 Hz and b) 5792 Hz model scale.
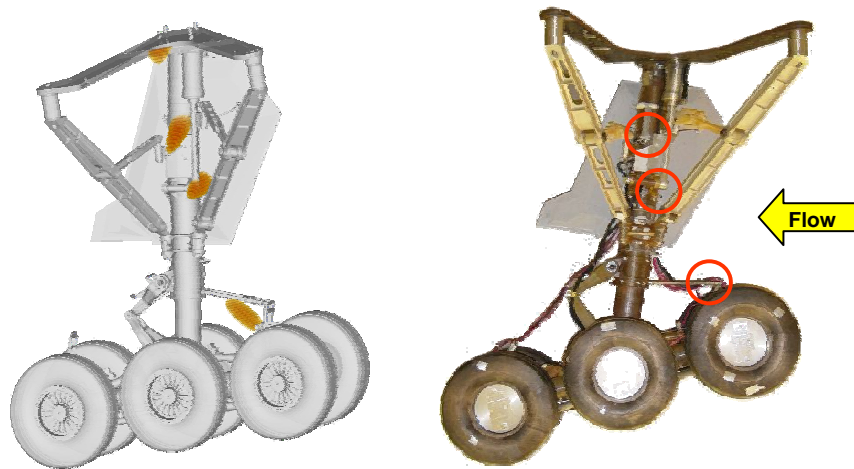

Figure 11: Beamforming results at band # 151 (6137 Hz center frequency, model scale).

7   CONCLUSIONS

A 32-element phased array system and all the computational code for the beamforming process have been successfully developed at Virginia Tech. Many preliminary tests were conducted to evaluate the capabilities of the array in out-of-flow, in-flow, reverberant and anechoic environments. The most significant results for these preliminary tests were presented. The phased array was used for noise source identification on a high fidelity 26%-

scale 777 aircraft main landing gear model. The most significant noise sources were clearly identified. Examples of the noise source identification on the landing gear model were presented. The tests were conducted at the Virginia Tech Stability wind tunnel.

8   ACKNOWLEDGMENTS

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