# ACOUSTIC HOLOGRAPHY APPLICATION FOR THE SOURCE IDENTIFICATION

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***Abstract.*** *Microphone-array-based methods are known as alternatives for noise source identification in vehicle interior, aircrafts, construction equipment, etc... In industrial environments as heavy machinery, the technique may be useful to take part on a predictive maintenance system by obtaining information not ready available by other traditional methods.*

*The goal of noise source identification is to identify, among the overall sound field produced, the most important sub-sources of an object in terms of their position, frequency contents and sound power.*

*In this work we will discuss the comparison between different holography methods along with parameter studies in order to better understand their advantages and limitations, helping select the technique which might be best suited to the application at hand, the predictive maintenance a mechanical system.*

*In the paper some applications of near-field acoustic holography (NAH) and phased array methods, will be discussed. Near-field acoustic holography is a measurement-based tool which visualizes the acoustic fields radiated by the noise emitting object, giving the engineer the noise source information required. Phased array methods are techniques of mapping noise sources by differentiating sound levels based upon the direction from which they originate. The method is very quick, allowing a full map to be calculated from a single-shot measurement.*

*These methods were implemented and applied to some noise sources in different environmental conditions.*

***Keywords****: acoustics, source identification, holography, measurements, microphone array*

## 1. INTRODUCTION

In the current practice of monitoring and diagnosis of equipment, as well as in the acoustic development of industrial products such as vehicles, a special interest is related to the location of noise sources in general. If one knows, at least approximately, the position of the most relevant sources the diagnosis can focus on the possible causes of noise at these specific locations on the equipment. It turns to be a complementary information along with the noise spectra to allow a correct identification of ongoing problems. In this context techniques which allow a deeper understanding of the acoustic field radiated appear as promising to deal with the problem of the sound source finding. Among these techniques the Acoustic Holography and the Beam-Forming have recently got special attention by many researchers. This work summarizes both techniques discussing their implementation and suitability to be used in a predictive maintenance program for a gas-turbine operating on a power-plant. Results obtained from both implementations are discussed along with the instrumentation developed.

## 2. ACOUSTIC HOLOGRAPHY

If one is able to know, through measurements, the sound field on a specified position, for instance over a defined surface, it may be possible to reconstruct the sound field in other portions of the space. The most common application of this idea is to measure the sound pressure in a measurement plane through an array of microphones and estimate the sound field in other planes, parallel to the first one (Maynard *et al.,* 1985) (Veronesi and Maynard, 1987). In order to accomplish this one must take into account the solutions of the wave equation governing the sound propagation in the space around the measurement plane (Kinsler *et al.,* 1982). Since we are supposing that we are only dealing with the propagation, and the actual source position is not known in advance, this solution is represented by the *Green*-Functions (Steiner, 1998) for a source-free region. Also important for the choice of the proper *Green*-Function is the type of sound field actually present in the measurements. A typical assumption is that they are done in a free-field.

Implementing this means that one will consider for some point in space a sum of the propagation of waves that originate on each microphone and, exhibit in their positions a complex sound pressure $P_n(\omega)$ , where *n* stays for the microphone index. The propagation must take into account the direction determined by the point of interest and the microphone, which lead to a expression of the wavelength by a vector with cartesian components in Eq. (1)

$$\vec{k}=[k_x, k_y, k_z] \quad . \tag{1}$$

Considering the time frequency in the harmonic solution of the wave equation a condition is imposed on the square norm of the wave vector $\vec{k}$ , such that Eq. (2) must hold:

$$\|\vec{k}\|^2=\left(\frac{\omega}{c}\right)^2 \quad . \tag{2}$$

where $c$ is the speed of sound and $\omega$ the time frequency of the harmonic wave under consideration.

The, plane, Holography is constructed by estimating the complex sound pressures on a specified plane, parallel to the measurement plane. Here it will be considered that the coordinate Z-axis establishes the direction of interest, i.e. planes are parallel to the X-Y-plane. Thus the estimated harmonic complex pressure on a given point of the estimate plane can be found by Eq. (3), using the first Rayleigh integral (Harris, 2005) equation:

$$p(\vec{r})=\iint p(r')G_d(r-r')dxdy \quad . \tag{3}$$

where, $G_d$ is known as the Dirichelet Green's function. The two-dimensional spatial Fourier transform to the wavenumber domain, represented for $P$, is used for its evaluation. Applying the convolution theorem to Eq. (3) yields Eq. (4):

$$P(k_x,k_y,z)=P(k_x,k_y,z')g_d(k_x,k_y,z-z') \quad . \tag{4}$$

where $z \geqslant z'$ and $g_d$ has a known closed form solution for the planar case given by Eq.(5):

$$g_d=\exp ik_z(z-z') \quad . \tag{5}$$

The space domain form of the pressure distribution is recovered by performing an inverse two-dimensional Fourier transform on the left hand side of Eq(4).

## 2. BEAM-FORMING

Beam-Forming is a method of mapping noise sources by assessing sound levels based upon the direction from which they originate. The implementation of the method is based on the delay-and-sum of the measured signals in each microphone. This can be accomplished either in the time or in the frequency domains (Brandstein, 2001). Firstly a time delay is determined based on the the locations of the array sensors and the propagation direction of the sound. This time delay is converted into a frequency dependent phase delay in the case of the frequency domain implementation. One key assumption made is that the incident waves can be regarded as plane waves, conversely it is assumed that the array is placed in the far-field of the sources.

The idea behind the technique is that a coherent sound, in the form of a plane wave, coming from a specified direction and being received by different microphones will lead to similar signals that are delayed on time based on the different travel paths. If one considers that the other sources, which may be present and are spatially dispersed, will not be coherent with the main source, under study, the mean value of the different microphone signals, delayed accordingly to their spatial location and propagation direction, will reflect only the coherent signal coming from that chosen direction (Christensen, 2004). The other signals being averaged close to zero.

Implementing this idea in the frequency domain can be expressed by Eq.(6):

$$B(\omega,\vec{r})=\frac{1}{N}\sum_{n=1}^{N}P_n(\omega)\exp\left(-j\omega\frac{\Delta_n(\vec{X}_n,\vec{r})}{c}\right) \quad . \tag{6}$$

where $B(\omega,\vec{r})$ is the complex sound pressure amplitude, with frequency ω, corresponding to incident waves from direction defined by $\vec{r}$ . The summation is done over all $N$ microphones, individually indicated by subscript $n$, for all measured complex sound pressures $P_n(\omega)$ . The different phase delays $\frac{\Delta_n(\vec{X}_n,\vec{r})}{c}$ can be seen inside the exponential and depends also on the vector position $\vec{X}_n$ of each microphone. The pure complex number is represented by $j$ and $c$ is the speed of sound. In Eq.(6) the phase delay is calculated from the different lengths of the incident wave paths $\Delta_n(\vec{X}_n,\vec{r})$ from microphone to microphone, related to a common reference position on the array. If we put the coordinate origin in this reference these lengths are found as in Eq.(7) as the simple vector projection of $\vec{X}_n$ into $\vec{r}$ :

$$\Delta_n(\vec{X}_n,\vec{r})=\vec{X}_n\cdot\vec{r} \quad . \tag{7}$$

Implementing these ideas with a generic description of the vectors $\vec{X}_n$ allows one to construct a mapping of the incident wave amplitudes. In the case presented here a coordinate system will be used for the Beam-Forming with a vertical Y-axis with the Z-axis pointing towards the source. The direction on the horizontal plane is the azimuth angle $\phi$ and the elevation angle $\theta$ represents the direction in the vertical plane. The graphic representation of $|B(\omega, \phi, \theta)|$ will show the directions with more incident sound energy.

Unlike the Holography the Beam-Forming does not reveals the nature of the sound-field in the surrounding of the array. It only enables one to find out from which main direction the sound is falling into the array, which is in turn frequency dependent. It also allows a "*real-time*" analysis of the incident sound, since it can picture a snap shot of the waves reaching the array. Of course this snap-shot is restricted by the amount of time necessary to acquire the samples needed for the FFT-algorithm being used.

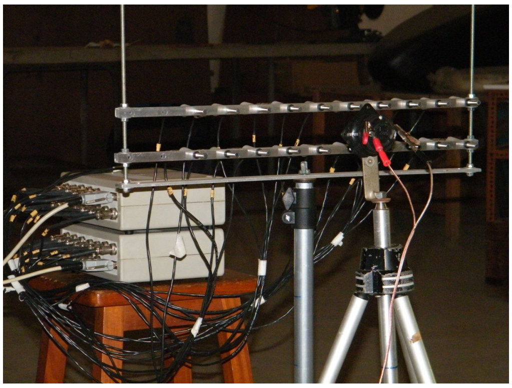
Since the formulation is generic the actual form of the microphone array, i.e. their spatial distribution, is of minor concern. The sole requirement imposed, in form of a spatial aliasing condition, is that the distance between microphones should guarantee that there will be at least two of them per wave length. Higher frequencies thus needing a closer spacing will need more microphones to correctly map the incident waves. Nevertheless an unstructured array with many sensors may allow a suitable choice of different microphone sets, depending on the frequency of interest, based on their spacing (Yunhong, 2005).

## 4. INSTRUMENTATION

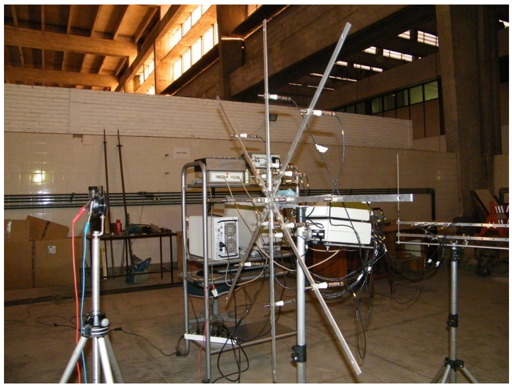
As shown in the previous sections one major drawback of both methods is the amount of microphones needed. Although the Beam-Forming can work with fewer sensors than the Holography the requirements on microphone spacing for higher frequencies again forces the use of a close spaced array. Therefore, in order to keep the overall cost of the instrumentation at acceptable levels, electret microphones were built by the Laboratory of Acoustics&Vibration itself, which are calibrated in amplitude and phase through an impedance tube adapted for that purpose.

The signal conditioning is also specially developed for this application and fit in an air-tight casing to be used in potentially hazardous environments. Using these microphones two kinds of planar arrays were studied, a linear array with two rows of ten microphones each and a circular array with three concentric sets of microphones.

Figure 1 shows the linear array with microphone spacing of 50mm in each direction and the circular array with microphones spaced by an angle of 45º and only two circles, at r=120mm and r=360mm, fitted with sensors.



a) Linear array



b) Circular array

Figure 1: Different array geometries

The arrays were tested using a loudspeaker in different positions in front of the arrays, radiating on a frequency range from 200Hz to 2kHz for the Beam-Forming and a fixed frequency of 2kHz for the Holography.

The holography estimates were created from repeated measurements with the array displaced vertically and horizontally in order to create a larger virtual grid. The sound sources radiated noise in steady-state and a reference microphone, fixed and close to the sources, was used to match the phases between measurements with different grid positions.

Measurement planes for the Holography, with the linear array, were chosen at distances of 100mm, 150mm and 500mm from the noise source, either loudspeaker or the mentioned equipment. For the Beam-Forming the same set of measurements were used. The signals were obtained inside the laboratory with low background noise but some reflections due to the obstacles around the array (furniture in general). The circular array was used on the outside of the laboratory where higher background levels were present.

A didactic equipment consisting of electric motor, mating gears, unbalance disks and rolling bearings, is also used to simulate a more complicated sound source in front of the arrays.

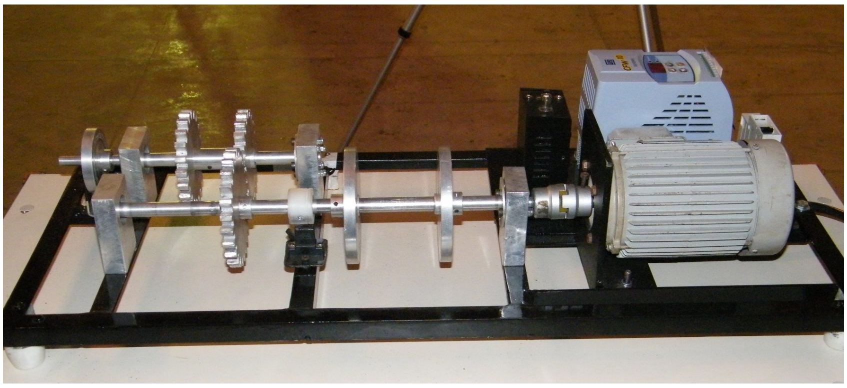
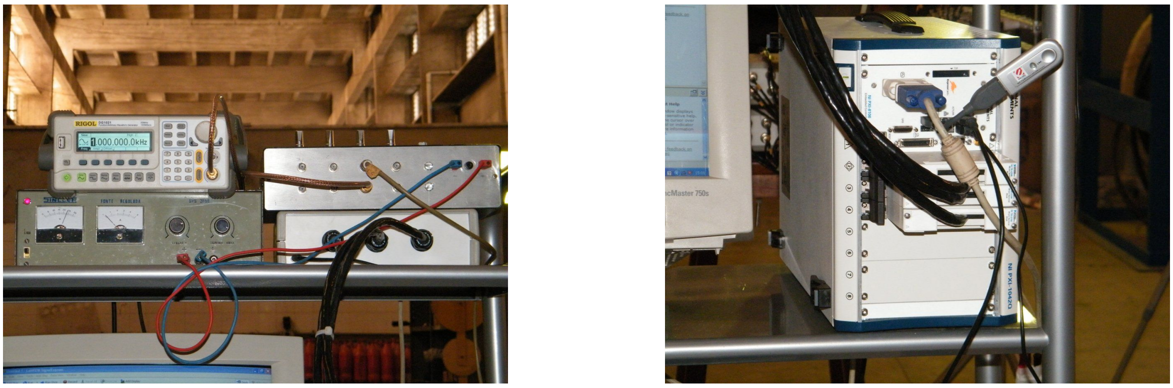


Figure 2: Equipment used for the experiments

The signals of twenty microphones in the linear array, plus one reference microphone, or of the sixteen microphones with the circular array, were acquired with a PXI-unit from National Instruments. The signal source for the sweep, the acquisition hardware and the amplifier can be seen in Fig. 3.



a) Signal generator, amplifier, power source and interface b) Acquisition hardware

Figure 3: Experimental setup with acquisition hardware

The signals were measured and saved in the time domain. Further processing, including transformation into frequency domain and the implementation of the holography and beam-forming, were done in MATLAB.

## 5. RESULTS

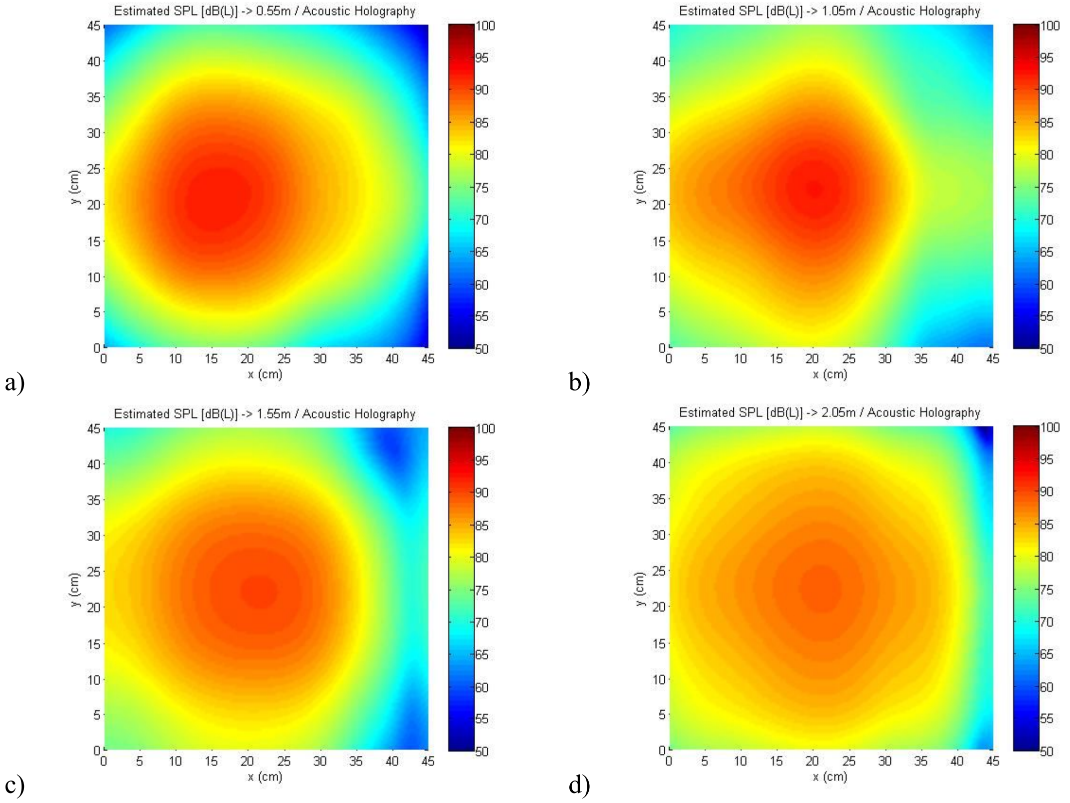
The estimated sound fields for the holography of a loud speaker radiating with 2kHz at a position below and to the left of the grid center are shown in Fig. 4 and for the didactic equipment in front of the array in Fig. 5.

In both cases shown the measurement plane was chosen at a distance of 100mm from the sources, in order to stay in the acoustic near-field.

The results obtained with the beam-forming are shown in Fig. 6 and Fig. 7 respectively for for a loudspeaker radiating a sweep between 200Hz and 2,0kHz in four different positions in front of the array and for the didactic equipment in two different positions.

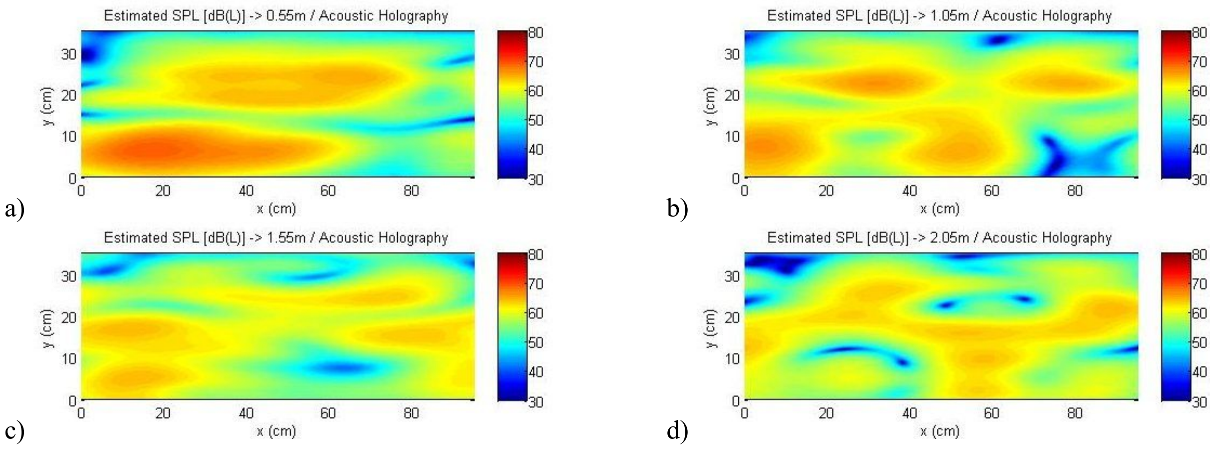
In the graphics shown in Figure 4 it can be seen that the closer you get to the source the estimated levels are higher as expected. Far away from the source the levels tend to a point source at the location of the loudspeaker, and the spreading of the acoustic waves can be recognized.

Figure 5, corresponding to a more complicated source, shows the different patterns of sound as emitted from the equipment in the approximate position of Figure 2. Close to the source the estimated levels are higher in the vicinity of the mating gears. When the estimation plane is positioned farther away from the equipment the sound pattern exhibits the influences of other sources too.




a)0.55m b)1.05m c)1.55m d)2.05m

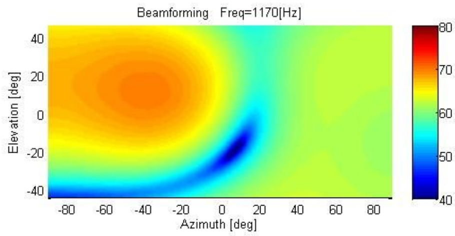
Figure 4: Estimated sound pressure levels at different planes – Loudspeaker @ 2kHz



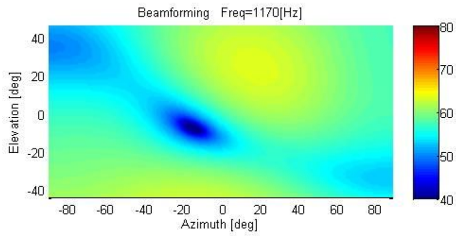

a)0.55m b)1.05m c)1.55m d)2.05m

Figure 5: Estimated sound pressure levels at different planes – Didactic Equipment @ 1.7kHz

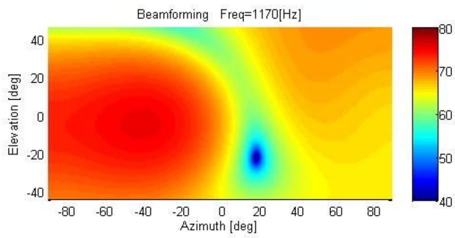
Figure 6 refers to different positions of the loudspeaker in front of the Beam-forming array, which can be recognized by the higher levels shown in red color. The same behavior can be seen in Figure 7 for the didactic equipment in front and at the right side of the array. Again the higher levels correspond to the position of the main source. The levels in Figure 7b are lower due to the greater distance to the array when compared to Figure 7a.



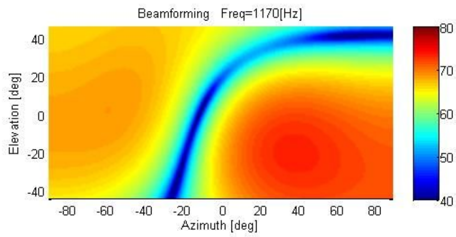

a)




b)



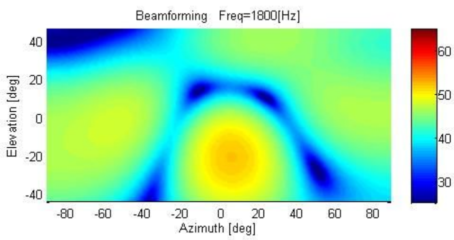

c)



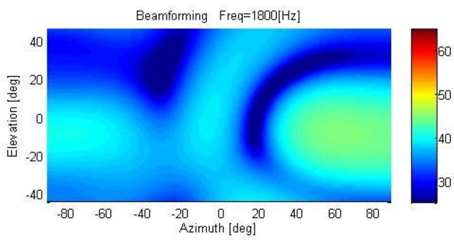

d)

a) Az. -45$^{O}$ / Elev. 18º b) Az. 0$^{O}$ / Elev. 18º c) Az. -45$^{O}$ / Elev. 0º d) Az. 45$^{O}$ / Elev. -11º

Figure 6: Beam-forming results for a loudspeaker in 4 different positions @1.17kHz




a)




b)

a) in front of the array – 0.8m b) at the right (45$^{O}$) from the array – 2m

Figure 7: Beam-forming results for a didactic equipment in 2 different positions @1.8kHz

## 6. INDUSTRIAL APPLICATION

The primary goal of this investigation is the application of the techniques as the core of a predictive maintenance system to observe the operating conditions of a gas turbine in a power plant. The instrumentation developed, specially its signal conditioning, is sealed in an hermetic enclosure to be able to operate in hazardous environments.

A major problem posed is the overall dimensions of the compressor and turbine sections of the equipment, as seen in Fig. 8.

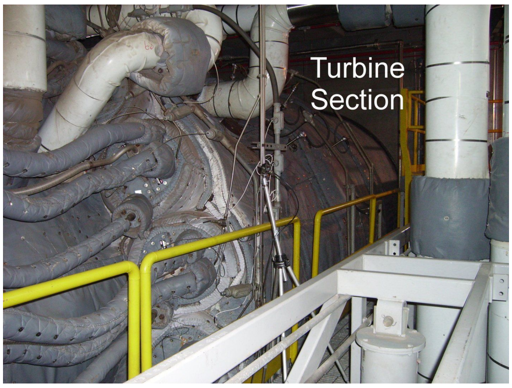



Figure 8: View of the trbine section of the gas turbine

The use of the holography will impose a very large array, whereas the beam-forming will give practical results with a smaller one. Figure 9 shows the noise spectra inside the turbine enclosure, showing the main frequencies of interest.

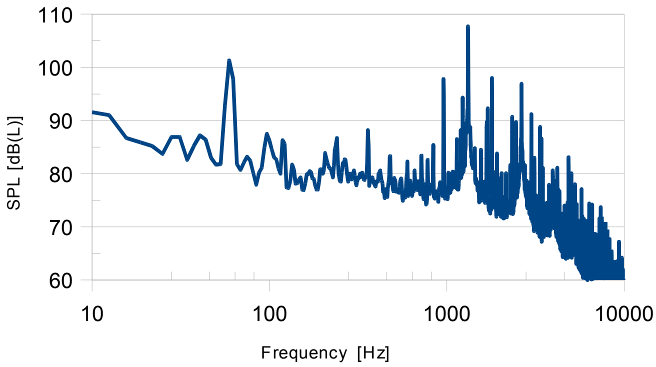



Figure 9: Noise inside gas turbine enclosure

It can be seen from Fig. 9 that the frequencies of interest ranges from 40Hz to approximately 3.5kHz. In order to maintain good resolution in a broad frequency band the microphone array will consist of a circle with radius from approximately 100mm and twelve microphones.

Two such arrays are being installed on the plant, one pointed towards the compressor section of the turbine while the other is directed to the turbine section, both built in the same instrumentation previously mentioned.

## 7. CONCLUSIONS

Both approachs, the Acoustic Holography and the Beam-Forming are powerfull tools that allows the engineer to better understand the nature of the sound-field based on a set of measurements. The major drawback is related to the amount of microphones needed to correctly map the sound field parallel to the measurement plane, in the case of the holography or the incident waves in the beam-forming.

The holography requires that the sound source of interest, which is not known in advance, must be completely spatially covered by the measurements, thus requiring a somewhat large plane depending on the application. On the other hand the beam-forming needs a larger number of sensors if one is interest in higher frequencies, due to the spatial aliasing effect. Other disadvantage is that the information is given in terms of the main directions of incidence of the sound waves. For sources not so close to the array it may lead to a poor source location, for small angular errors translates in bigger position errors for larger distances.

The industrial application of the techniques in the case of the predictive maintenance of a Gas-turbine will suggest the beam-forming since the overall dimensions of the equipment will require a large measurement plane, increasing the instrumentation costs. Since the application will be based on the time variation of the incidence pattern on the array further work will be dedicated to the establishment of comparisons between these patterns and to extract usefull information to be related to the turbine condition. The implementation of the techniques were successful and a suitable and cost effective instrumentation is developed, which is able to be used in hazardous environments.

## 8. ACKNOWLEDGEMENTS

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