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Extended summary

Noise Source Identification:

an innovative approach

of the Selective Intensity method

Curriculum: IngegneriaMeccanica e Gestionale

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Abstract.

Noise control research field has grown widely in the last thirty years. That has led to the development of several noise source identification techniques. Unfortunately, even though they perform very well when applied in free field conditions, they still lack of accuracy when applied in reverberant/really noisy environments.

This thesis deals with a new approach of the noise source identification method named Selective Intensity (SI), appositely developed to operate in highly reverberant environments like aircraft cabins. The Selective Intensity is a vibro-acoustic correlation technique capable of locating the strongest structural sources that mostly radiate to the acoustic field. The innovation behind the proposed approach consists in involving, for the very first time, the usage of a Scanning Laser Doppler Vibrometer (LDV) and a Sound Intensity p-p probe. The optical non-contact nature of the LDV lets to collect information with a very fine spatial resolution and not modifying the dynamic behaviour of the analyzed structure. For the first time a comprehensive analysis of the numerical aspects lying behind the SI is performed. Moreover a deep sensitivity analysis is undertaken, focusing on those parameters that mostly influence the SI outcomes (e.g. sources coherence). Results from real applications of the technique are widely presented, but particular attention is given to those obtained on an Alenia Aeronautica ATR42 ground test facility.

The possibility of further separate structural and acoustic sources contributions from data measured for the Selective Intensity calculation would therefore represent a useful goal to achieve. Based on these considerations, the exploitation of the Independent Component Analysis (ICA) technique for vibro-acoustic purposes is also handled throughout the thesis. A debate about ICA merits and limits is presented; moreover, performances of that approach in achieving source contributions reconstruction are described regarding both simulated and real test cases. Results obtained demonstrate the technique is worth of attention and it deserves further studies, even though it suffers of some drawbacks.

Keywords.

Selective Intensity, Laser Doppler Vibrometer, Noise Source Identification, Regularization Methods, Independent Component Analysis

1 Problem statement and objectives

The first natural step in facing a noise control problem consists in locating areas of strongest acoustic emission. Several noise sources identification tools are available up to now. They are either purely structural or acoustical. Accelerometers and Laser Doppler Vibrometers (LDV) [1] belong to the first category and they are extremely useful sensors for measuring vibrations. They both provide information directly measuring on the structure surface, but their main difference is due to the non-contact nature of the LDV, which makes it exploitable in a wider range of applications, since it does not modify the structure dynamics. From the purely acoustic side, the Sound Intensity technique [2][3] has represented (and sometimes still does) the reference measurement methodology for sound power levels determination and noise source localization for several years, taking the advantage, compared to microphones, from the vector nature of the Sound Intensity. During the last twenty years some new techniques have grown importance as noise source identification tools. Near-field Acoustic Holography (NAH) [4][5] and Beamforming [6] use array of microphones in order to measure the sound field simultaneously. In the last 10 years a new sensor have come up capable of measuring both air particle velocity and air pressure: the Microflown *p-u probe* [7][8][9]. Unfortunately the *p-u probe* presents some issues in terms of phase mismatch at low frequencies, and the very-near-field approximation it is based on is still under analysis by the scientific community.

Even though the above mentioned acoustic techniques perform very well in free-field conditions, they still suffer when applied in highly reverberant environments. Several steps have been done in the last two years in improving their performances, like for example the usage of a double layer antenna [10] for NAH calculations or the Average Beamforming approach [11] for interior 3D sound sources localization, but they both were still at the development stage at the beginning of this thesis. Moreover the high frequency nature of Beamforming makes it not suitable for having a fine spatial resolution at lower ones, while costs linked to the usage of a NAH double layer antenna are high, considering it is composed of more than a hundred ¼ inches microphones. Because of these issues, the need for developing a measurement technique capable of correctly identifying noise sources on a specific system in its working environment is becoming more and more pressing and industries hardly push towards that direction.

The aim of this PhD project is therefore to develop a measuring methodology to indentify noise sources in those complex acoustic fields characterized either by a high reverberant component or by a strong background noise and able to supply outcomes with a fine spatial resolution. Separating contributions coming from each identified source would also represent a really useful outcome for engineers dealing with noise control problem, because it would give the possibility of better understanding the system under analysis from a *wide vibro-acoustic point of view*. For that reason, a further source separation approach was started within this PhD project, involving the computational procedure named Independent Component Analysis (ICA) [12].

2 Research planning and activities

This research program aims to develop a measurement tool in order to fulfill the Noise Source Identification task in complex acoustic environments and, eventually, perform a further Source Contribution Separation.

The main hypothesis lying behind the Noise Source Identification process consists in the observation that, while extraneous noise sources spoil someway acoustic data, they do not influence vibration ones, which therefore constitute a coherent estimator of the true sources emission. That obviously holds if those extraneous sources do not themselves contribute in dynamically exciting the structure under analysis. Upon this assumption, the idea is to measure structural (vibrations) and acoustic data and afterwards correlate them in order to exclude unwanted components such as reverberating ones. This approach represents the rationale behind the Selective Intensity (SI) technique. This measurement method lets to both identify the strongest noise sources which are somehow related to the structure vibrations and, at the same time, to estimate the acoustic contribution of each source, as if it would act separately from the others.

The main innovation of the proposed method, compared to previous existing ones [13][14], consists in using, for the very first time, a Scanning Laser Doppler vibrometer (SLDV) to collect vibration information. The combined usage of the SLDV and a standard sound intensity *p-p probe* (exploited for acoustic data recording), simultaneously moved along the measurement area in order to perform a scan of the system under analysis, allows to sample the structure really accurately, thus letting to identify noise sources with a very fine spatial resolution.

The physical system is modeled as a Multiple Input Multiple Output (MIMO) system, where the inputs are the vibration velocities and the outputs the pressures measured by the microphones of the *p-p probe*. Since n vibration sources and n intensity values are measured in front of the surface, the reconstruction of a Selective Intensity map with n points is possible, thus letting to spatially locate the main noise sources. The linear relations linking the inputs and the outputs of the model can be represented as Frequency Response Functions (FRFs), which can be evaluated solving the problem shown in Equation (1).

$$\mathbf{G}_{\mathbf{vm}} = \mathbf{G}_{\mathbf{vv}}\mathbf{H} \tag{1}$$

$\mathbf{G}_{\mathrm{vv}}$ is the n by n cross-spectral matrix of vibration velocity data, $\mathbf{G}_{\mathrm{vm}}$ is the n by 2 cross-spectral density matrix between the vibration velocities and the acoustic pressure at the two microphones and $\mathbf{H}$ is the n by 2 transfer function matrix (it has to be reminded that the column size 2 is due to the number of microphones used in the sound intensity probe).

Once the FRFs are calculated the microphones cross-spectral density, whose imaginary part is usually exploited in standard sound intensity evaluation, is replaced by a so called "Conditioned Cross-Spectral Density" (CCSD), according to Equation (2) (here shown for source *l*).

$$G_{m_1m_2,vl}(f) = H^*_{l1}(f)H_{l2}(f)G_{v_lv_l}(f) \tag{2}$$

The CCSD is then used in the Selective Intensity estimation according to the Cross-Spectral Sound Intensity approach [2].

The scanning approach, exploitable only if the observed phenomenon is stationary, implies measurements to be performed in a sequential way. That requires the usage of a phase realignment procedure in order to compensate for sensor phase mismatch and to avoid the underestimation of the vibration cross-spectral matrix. The phase realignment can be achieved considering a phase reference sensor properly placed on the structure under test (that is not for example on a nodal line) and introducing a *modified phase* according to Equation (3).

$$\begin{aligned} \angle G^r_{vivj}(f) &= \angle FRF_{vj/R} - \angle FRF_{vi/R} \\ \angle G^r_{vim1_i}(f) &= \angle FRF_{vi/R} - \angle FFT_{m1_i} \\ \angle G^r_{vim2_i}(f) &= \angle FRF_{vi/R} - \angle FFT_{m2_i} \end{aligned} \tag{3}$$

where $\angle FRF_{vi/R}$ is the phase of the FRFs between the vibration velocity at source *i* and the signal (*R*), taken as reference for the vibration measurements, and $\angle FFT_{ml_i}$ is the phase of the acoustic pressure spectrum measured by microphone *l* at source *i*. Under the hypothesis of stationary phenomenon, this allows to make all the vibration inputs in phase with the same reference and thus to align in time all the measured signals in order to avoid errors due to time delay between the vibration sources. Considering that the sound pressures measured by the two microphones of the *p-p probe* are acquired simultaneously to the vibration velocity corresponding to the same position, the acoustic pressure signals do not need realignment, and therefore relative phase between microphone and phase reference transducer (*R*) does not appear in the second and third equation of Equation (3).

The increase in the measurement points lets to localize the noise sources with a fine spatial resolution. Unfortunately that also represents an issue form a numerical point of view because of the size of the inverse problem which has to be solved. Numerical instabilities in the solution of Equation (1) are mainly linked to the ill-posed nature of the vibration cross-spectral matrix. If two measurements points are set too close to each other they do represent the same information, thus not contributing to the physical rank of the cross-spectral matrix and thus giving rise to ill-conditioning during the inversion. The complexity of the mode shapes can also be an issue. The $\mathbf{G}_{vv}$ can be seen as a result of uncorrelated modal coordinates each corresponding to a couple of different singular values and eigenvectors. When the mode shape is complex there are small singular values of the decomposed cross-spectral matrix that contribute to the mode and also the off-diagonal terms get stronger (the matrix becomes sparser): this can lead to ill-conditioning during the inversion. It is quite straightforward that noise on the laser measurement is also an issue, since it masks the true information. When the vibration cross-spectral matrix is ill conditioned the solution obtained through the Singular Value Decomposition (SVD) is subjected to numerical instabilities, because the small singular values, when inverted, contribute the most. One way of avoiding these unwanted amplifications is to exploit the Tikhonov Regularization method, which let to rewrite the solution according to Equation (4).

$$\mathbf{H}_{\alpha} = \left(\mathbf{G}_{\mathbf{vv}}^{H}\mathbf{G}_{\mathbf{vv}} + \alpha\,\mathbf{I}\right)^{-1}\mathbf{G}_{\mathbf{vv}}^{H}\mathbf{G}_{\mathbf{vm}} \tag{4}$$

where $\mathbf{I}$ is the identity matrix and α the regularization parameter. Computing the SVD, the regularized solution can then be rearranged as in Equation (5).

$$\mathbf{H}_{reg} = \mathbf{V}\mathbf{\Sigma}_{reg}^{-1}\mathbf{U}^{H}\mathbf{G}_{\mathbf{vm}} = \sum_{i=1}^{n}\frac{\sigma_i^2}{\sigma_i^2 + \alpha_i}\frac{\mathbf{u}_i^H\mathbf{G}_{\mathbf{vm}}}{\sigma_i}\mathbf{v}_i \tag{5}$$

$\mathbf{U}$ and $\mathbf{V}$ are the matrices of left and right singular vectors $\mathbf{u}_i$ and $\mathbf{v}_i$ of the matrix $\mathbf{G}_{vv}$, $\mathbf{\Sigma}$ the singular values (σ_i) matrix, α the regularization parameter. The superscript H denotes the Hermitian transpose. If a high regularization coefficient is chosen, the Tikhonov regularized solution will act as a sharp low-pass allowing the components with low indexes. The transfer function matrix $\mathbf{H}_{reg}$ will take into account just for these components, not considering for example the complexity of the vibrational behaviour if a complex mode shape occurs at the particular frequency at which calculation is performed. On the other side, if the α value is too small, the transfer function could suffer because of noise on the vibration measurements (showing as small singular values), thus assigning wrong strengths to the reconstructed sources. It comes natural the best solution is the one that balances between the two situations, that is neither loosing information nor suffering too much because of sensor noise. Some automatic procedure for the regularization parameter selection (e.g. Generalized Cross Validation, L-curve criterion) should therefore taken into account in order to choose the best α.

Concerning the Source Contribution Separation task, the leading idea consisted in trying to exploit a computation technique named Independent Component Analysis for vibro-acoustic problems. ICA [12][15] is a statistical and computational technique for discovering hidden factors from sets of observed multivariate data (random variables, measurements or signals). It states a generative model of those data variables, which are assumed to be linear or nonlinear mixtures of some unknown latent variables. The main peculiarity is that the mixing system is also unknown. The latent variables are assumed to be nongaussian and mutually independent: that is the reason they are called independent components. ICA tries to estimate those independent components (often named ICs).

When applied to the source separation field ICA is often named as Blind Source Separation. The term "blind" means that no information are known about the mixing process (it has to be estimated blindly). The first goal of ICA is to evaluate this mixing process.

In some simple mixing models each recording consists in a weighted sum of the source signals. In acoustics, because of multiple paths and multiple reflections of the acoustic waves, the mixtures are weighted and delayed. The sum thus become a filtered sum of the different sources, and the mixing system is called *convolutive*.

When dealing with vibro-acoustic source separation one has to understand the nature of the convolutive problem. The mixing model is linked to the modal response of the structure when considering vibration data (the mixing matrix is actually the FRF matrix of the object under test). That holds if no other external sources are acting on the structure, also contributing on its vibrational response. When acoustic data are considered (pressure recorded by microphones or reconstructed from Boundary Element simulations) the mixing matrix is a combination of the vibrational mixing matrix and the acoustic coupling within the sound propagating medium. If acoustic data are collected in the far-field in a reverberant environment, reflections from the walls also contribute to create the mixing model, thus making the solution of the convolutive problem much harder. If near-field measurements are considered, reflections from the environment do not represent an issue, since the direct field is much stronger than the incident (due to reflections) one, but the evanescent component of the radiated sound field can represent a further complication for BSS.

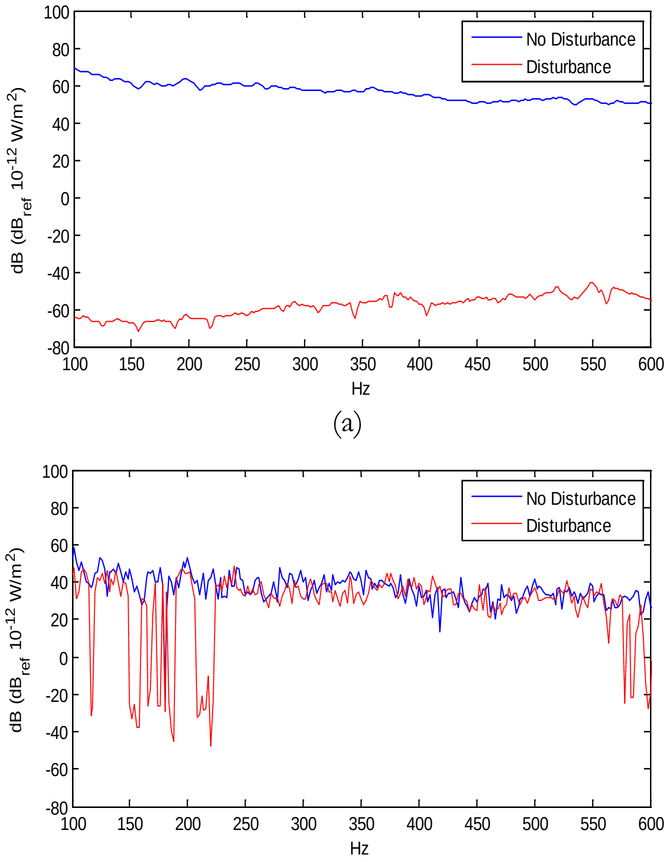
The approach developed in this thesis is based on a modified version of the CICAAR (Convolutive ICA based on Auto-Regressive model) algorithm [16]. This novel methodology involves the squaring of the ICA convolutive problem (when dealing with the overdetermined case – more sensors than sources) and the evaluation of the sources overlapping frequency ranges through the calculation of the Singular Value Percentage Contribution [17] of the sensors cross-spectral matrix. That enables the application of a subband decomposition approach [18] within ICA, letting to improve performances of the Source Contribution Separation step.

3 Analysis and discussion of main results

The Selective Intensity technique aims to locate noise sources in those complex acoustic environments characterized either by strong background noise or by long reverberation time. The combined usage of vibration and acoustic information lets to overcome those limits which are typical of a standard technique like the Sound Intensity, that last providing inconsistent results when applied in diffuse fields.

The capability of the proposed approach to fulfil the noise identification task is presented at first for a laboratory test where the measurement environment is spoiled through an artificially created background noise. The structure under test was a chipboard panel placed in the aperture of the anechoic chamber of UNIVPM, in order to have a controlled acoustic field in the measurement volume. The direct excitation was produced by a loudspeaker placed outside the measurement room and fed by white noise in the range of 1÷1200 Hz. The reverberating field has been simulated by adding a background noise with a second loudspeaker inside the anechoic room. The two loudspeaker produced an overall SPL of respectively 110 dB and 105 dB at 1m distance.

Figure 1 shows the comparison of results obtained from the Standard Acoustic Intensity and the Selective Intensity when measuring on the panel both in presence and absence of the simulated background noise.



(a)

(b)

Figure 1. Standard Sound Intensity (a) and Selective Intensity (b) with (dotted line) and without (solid line) disturbance

The Standard Sound Intensity in case of disturbance is completely negative, the sound intensity probe being closer to the disturbance speaker located behind it. On the other hand the Selective Intensity calculated in the conditions with and without disturbance almost overlaps: in the range 234Hz÷560Hz the overall difference is of 2dB: these values indicate that the process is able to remove the disturbance effect on the acoustic field. Some amplitude differences can be attributed to the augmented vibration field due to the background noise. The spikes evident in the Selective Intensity spectrum measured in simulated reverberating conditions are due to a decreasing of the conditioning of the coefficient matrix because of the presence of additional noise.

The Selective Intensity technique was also applied for the first time in a highly reverberant environment as an aircraft cabin. Results shown hereafter refer to a measurement campaign performed on an Alenia Aeronautica ATR42 full scale ground test facility in Pomigliano d'Arco (Naples, Italy) during the European project CREDO (Cabin noise Reduction by Experimental and numerical Design Optimization, 2006-2009).

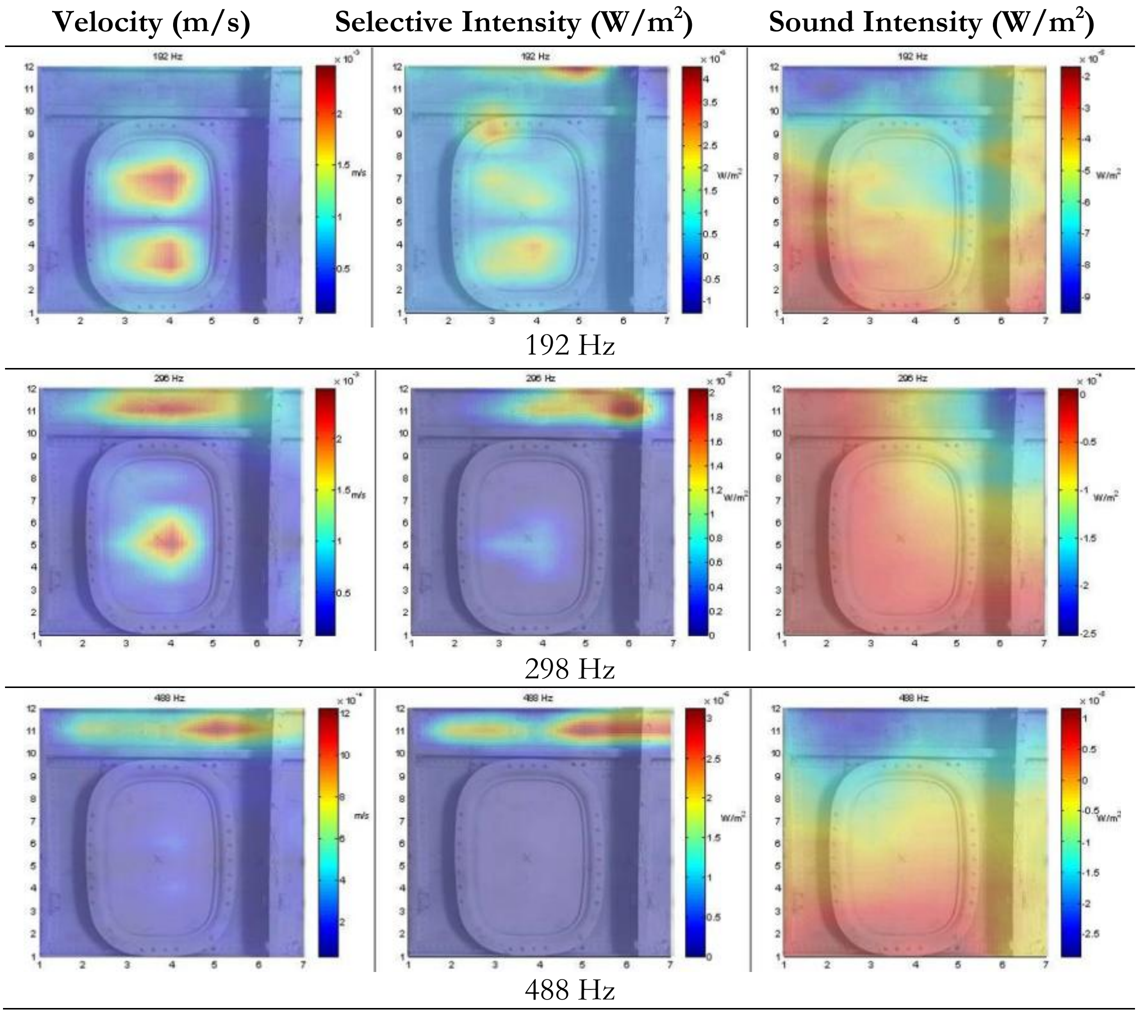


Figure 2. ATR42 ground test: Vibration velocity, Selective Intensity and standard Sound Intensity distribution at different frequencies (structural and acoustical excitation active)

Figure 2 shows Vibration, Selective Intensity and Standard Intensity maps at different frequencies evaluated when the fuselage section is excited both structurally and acoustically. It can be easily noticed that Standard Intensity suffers in correctly locating the strongest sound sources due to the diffuse nature of the sound field inside the aircraft cabin. On the contrary the Selective Intensity correctly identify those sources: the window frame and the upper-right part of the structure (the part directly excited by an electromechanical shaker).

The importance of the novel Regularization approach is also proved in Figure 3, where results obtained performing or not numerical regularization are presented.

Selective Intensity (W/m²)

Tikhonov regularization + GCV method	**No regularization**

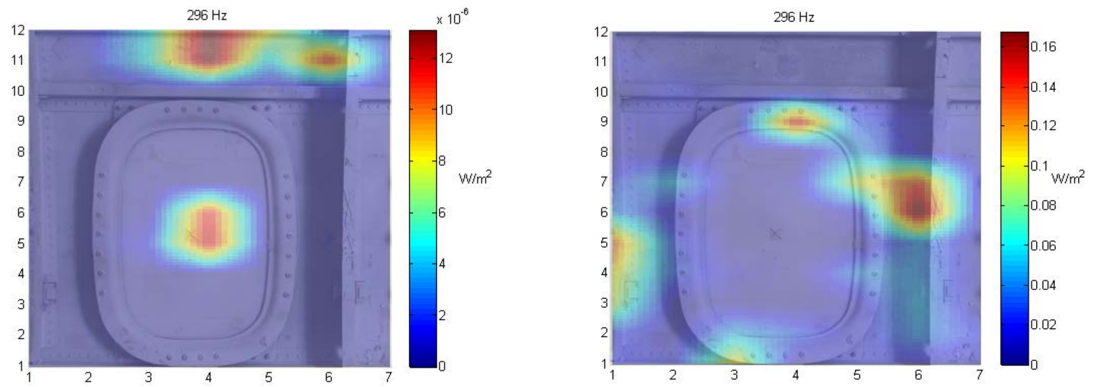


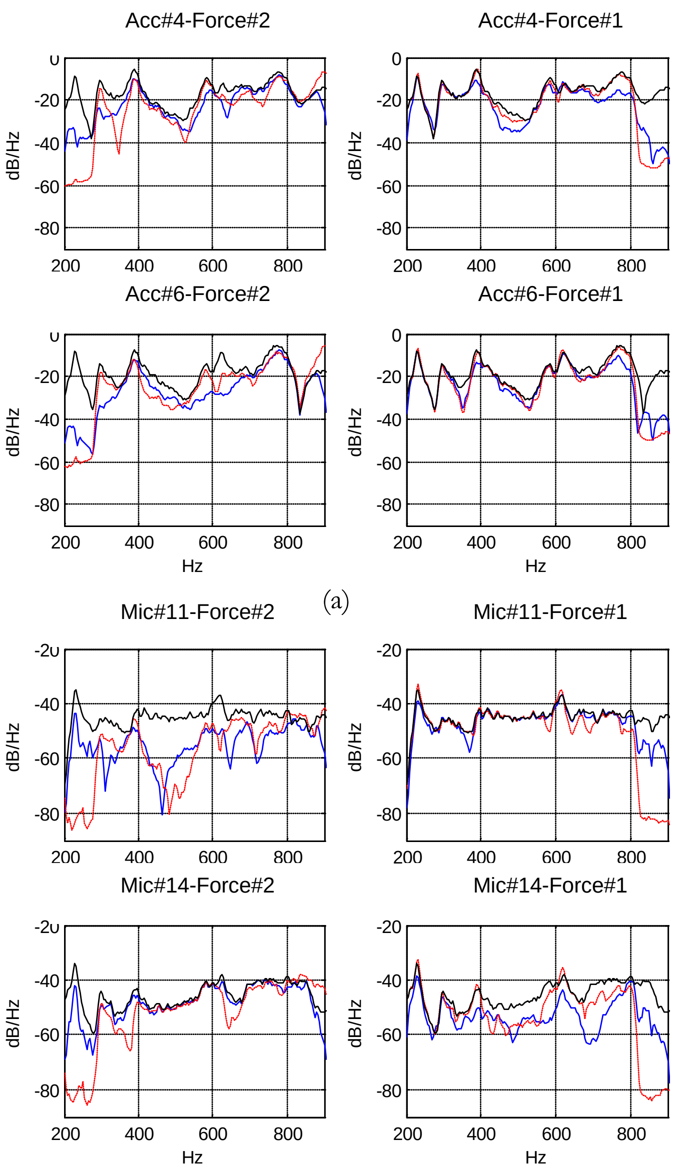
Figure 3. ATR42 ground test: Selective Intensity maps at 296 Hz obtained with and without regularization (case of shaker side excitation)

The investigations about convolutive ICA carried out in the thesis are based on both numerical simulations and experimental tests. For synthesis purposes only those last are presented hereafter. Measurements refer to tests performed by the author in the Experimental Area of Brüel & Kjær Sound & Vibration (BKSV) in Nærum, (Denmark). In order to fully test the MCICAAR (Modified CICAAR) capabilities of correctly reconstructing contributions due to different forces, acting simultaneously on the structure under analysis, both vibrational and acoustical data are taken into account, therefore both acceleration and pressure signals are processed by the algorithm.

Figure 4 shows acceleration (a) and pressure (b) components (Power Spectral Density evaluated in dB [dB_{ref} $10^{-5}m/s^2$] on 4Hz spectral resolution bandwidth), due to each acting force, in the frequency range 200Hz÷900Hz, that is the two forces overlapping frequency range. The black line represents the combined response sensed on each transducer when the two forces are acting simultaneously. The red lines stands for the acceleration responses of the plate when it is driven just by one force (Force#1 or Force#2): this responses were obtained alternatively switching off the shakers used to excite the structure under analysis. The blue lines symbolize the estimated acceleration components through the MCICAAR algorithm.

Regarding acceleration data it can be seen MCICAAR performs quite well at those frequencies where only one force is acting (below 300Hz and above 800Hz): an approximate reduction of almost 25dB/Hz (in terms of acceleration levels) is obtained.

Concerning pressure data the trend of the MCICAAR outcomes generally resembles the one of the true components. This is really evident for the anti-resonance at about 480Hz, which is characteristic of the contribution due to Force#2 on pressure at Microphone#11: that is not registered at all by the microphone when the two forces are acting simultaneously. On the other side it is possible to notice a worse performance, at least compared to the one from acceleration data, outside the overlapping frequency region of the two forces (below 300Hz and above 800Hz). The algorithm is not capable of completely removing the resonance at 250 Hz (which is due only to Force#1), thus resulting in a wrong reconstruction. Discrepancies are reduced at a maximum of 10dB/Hz within the region 300Hz÷800Hz.



(b)

Figure 4. - ICA reconstructed components (blue) compared to true components (red) and mixed signal (black) for vibration (a) and pressure (b) data

4 Conclusions

The aim of this thesis has been to develop a Noise Source Identification measurement technique capable of correctly locating sound sources in **complex acoustical fields**. The purpose rises from the market increasing demand of performing acoustic measurements on systems working in their operating environments, which could be either spoiled by a strong background noise or characterized by a long reverberation time.

The main innovations introduced within the SI approach can therefore be summarized as following:

- From a strictly methodological point of view one innovation consists in involving, for the very first time, a **Scanning Laser Doppler** and a **Sound Intensity probe,** simultaneously measuring on the same position and moved together in order to perform a scan on the system under analysis. Thanks to that procedure, more information about the structural sources behaviour can be collected, that allowing to **identify noise sources with a very fine spatial resolution**.
- The scan procedure requires two conditions to be satisfied: the analyzed phenomenon to be stationary and the usage of a phase reference sensor. **The importance of phase realignment**, in order to virtually change sequential measurements into simultaneous ones, has been widely discussed for the first time within the Selective Intensity procedure.
- If from one side the increasing of the number of investigation points leads to a more accurate spatial resolution in terms of source localization, from the other one it can severely put the technique capabilities to the test. That is because the SI calculation involves the **solution of an inverse problem**, which is often ill conditioned. For the very first time **numerical regularization** is considered in SI problems to tackle instabilities.
- For the first time a deep **sensitivity analysis** of the SI approach has been undertaken and **real application cases** illustrated.

Concerning the Source Contribution Separation task it has to be said ICA definitively represents an interesting approach. The SI involves the measurement of both vibration and pressure data: ICA has therefore been tested on the two physical quantities in a separated way, in order to understand the different performances of the technique in treating them. That is meaningful since different mixing matrices are generally involved behind vibration and pressure data (structural FRFs for vibration and structure-fluid transfer functions for pressure ones). It has been demonstrated MCICAAR leads to interesting results, that justifying the need for further studies in order to improve the accuracy of the approach. The "subband decomposition" step seems to be really promising concerning its capability of increasing the algorithm performance (especially in terms of a future noise control tool development); also, at least according to the author knowledge, considering that it has never been applied before to CICAAR.

Research fields are open within both the approaches, and the leading theme should be kept on their interaction, in order to provide a tool for a comprehensive noise control problem analysis.

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