Buenos Aires – 5 to 9 September, 2016
Acoustics for the 21st Century…

PROCEEDINGS of the 22nd International Congress on Acoustics

Materials for Noise Control: Paper ICA2016-882

Characterization of mufflers

Ahmed Allam [(a)] and Tamer Elnady [(b)]

[(a)] Ain Shams University, Egypt, a.allam@eng.asu.edu.eg

[(b)] Ain Shams University, Egypt, tamer.elnady@eng.asu.edu.eg

Abstract

Mufflers are widely used to reduce the exhaust and intake noise of fluid machines for different applications. Every muffler has to be tailored carefully to the engine to which it is connected. One very important tool of muffler design is the measurement of its properties, acoustic performance and pressure drop. Introduction of flow is a key issue as it simulates a real engine situation. The two source technique has been proven to be the most stable and most efficient technique to characterize the full scattering matrix of the muffler. At several companies and research institutes, this has become a standard measurement which is repeated frequently throughout the design process. This paper describes a new platform to measure the passive and active (flow-generated noise) properties of mufflers. Stepped sine excitation is used with simultaneous excitations from both sides of the muffler. The stepped sine excitation is optimized to reduce the needed time without jeopardizing the quality of the measurement. Measurement of flow background noise, microphone coherence, and pressure drop are also performed. This platform is based on a combined JAVA/NI software and National Instruments Data Acquisition cards, to automate the measurement accounting for different theoretical and practical considerations.

Keywords: Mufflers, Two-source method

Characterization of mufflers

1 Introduction

Mufflers are devices used to reduce noise emitted from different fluid machines. The first step of muffler design is usually to analyse the sound propagation in the duct networks connected to the intake and exhaust of the machine to determine the noise frequency spectrum. The muffler is then designed to suppress the frequencies contributing the most to the noise produced by the machine. Finally, the properties of the designed muffler are validated experimentally. This process is repeated whenever a new fluid machine is developed or its design is modified.

The two source technique is one of the common techniques used to experimentally evaluate the acoustic properties of mufflers [1]. It involves acoustically exciting the muffler once from each side (upstream and downstream). The excitation type could be broadband white noise excitation, or using stepped sine excitations. Pedrosa et al. [2], [3] suggested a method to use simultaneous upstream and downstream white noise excitation so that the measurement is done in one step. This reduces the time needed to perform the measurements and reduces the errors caused by the change of the measurement conditions between the two steps.

In this work, we introduce a new platform for automating experimental evaluation of the acoustic properties of mufflers. The new platform includes the implementation of a new algorithm for performing stepped sine two source measurements with simultaneous upstream and downstream excitations and reduced single excitation acquisition time. The new algorithm is validated by comparing its results to analytic and standard white noise measurements of different samples. The results are also compared to a conventional acquisition algorithm [4] implemented in the commercial software SIDLAB. A significant reduction in the acquisition time is observed without any noticeable effect on the quality of the measurement.

2 Theory of two source technique

The acoustic two-port theory is often used to analyse low frequency sound propagation in duct networks. The network is divided into a connected group of one dimensional two-port elements. Choosing the travelling waves pressure amplitudes (p_+, p_-) as state variables (Figure 1), the acoustic wave propagation inside an acoustic element with ports (a, b) can be described by the scattering matrix ($\boldsymbol{S}$) formulation:

$$\begin{bmatrix} p_{a+} \\ p_{b+} \end{bmatrix} = \begin{bmatrix} S_{11} & S_{12} \\ S_{21} & S_{22} \end{bmatrix} \begin{bmatrix} p_{a-} \\ p_{b-} \end{bmatrix} + \begin{bmatrix} p_{a+}^s \\ p_{b+}^s \end{bmatrix}$$

where p_{a+}^s, p_{a-}^s are the sound pressure waves generated inside the element and propagating through ports *a* and *b* respectively.

The elements of $\boldsymbol{S}$ and p_+^s must be determined in order to fully characterize an acoustic element. The two source technique is usually used to experimentally estimate the elements of $\boldsymbol{S}$. It incorporates the placement of the element (muffler) to be characterized between two pipes as shown in Figure 1. Each pipe is fitted with at least two microphones flush mounted to its inner

walls. The microphones are used to decompose the plane sound waves traveling in the pipe in the forward and backward directions. Two sound sources (speakers) are placed at the upstream and downstream ends of the pipe. The speakers are used to provide upstream and downstream sound wave excitations. By exciting the target muffler once from each direction and estimating the transfer functions between the microphones, the four elements of the scattering matrix of the muffler can be estimated [5]. The details of this procedure and the practical considerations are explained in details in the E2611-09 ASTM standard [6]. Knowing the elements of $\boldsymbol{S}$, the pressure amplitudes of the waves generated inside the element (flow generated noise) $p_{\mp}^{S}$ can then be estimated using the procedure explained in details by Lavrentjev et al. [7].

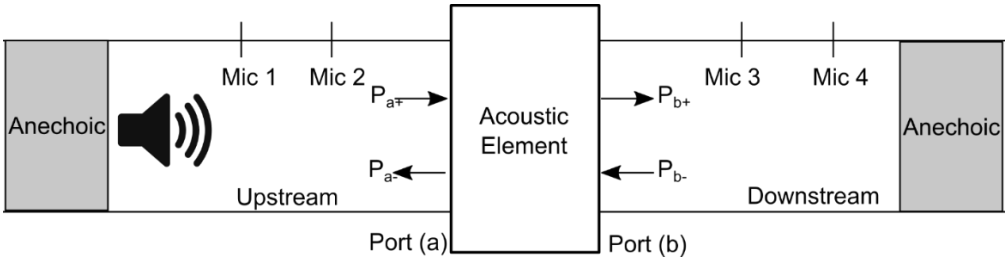

Figure 1: The Two-source setup and the state variables for the scattering matrix formulation.

3 Development of the new platform

The suggested two source measurement platform consists of four parts; the data acquisition system which is connected to the two source setup, the acquisition logic, the measurement logic and the Graphical User Interface (GUI). The GUI is used for facilitating the setting of the measurement parameters (microphone spacing, tube diameter, frequency range … etc.) and post processing of the estimated properties (viewing, plotting, exporting … etc.). The interface is developed using JAVA programing language for its GUI capabilities and the ease of its integration with the other parts of the platform. The inputs in the GUI are then sent to the acquisition logic. The acquisition logic is responsible for the excitation of the speakers via the output channels and collecting the data from the microphones using the input channels of the data acquisition system. It is then responsible for converting them into transfer functions. It is written using the ANSI C programming language, and uses NI LABWINDOWS/CVI which is an integrated development environment that provides built-in hardware libraries and analysis functions. It is capable of detecting and driving any appropriate NI data acquisition system that is compatible with the NI-DAQmx driver software and contains at least two output and eight input analogue channels. The choice of a low level programming language like C allows for faster data processing and low level hardware manipulation and optimization. It also allows for an overall small program size and fewer dependencies on external libraries. The last part, which is the measurement logic, is responsible for estimating the components of the $\boldsymbol{S}$ matrix from the transfer functions estimated by the acquisition logic. It also calculates other derived acoustic properties such as the transmission loss, insertion loss, flow generated noise … etc. The measurement logic is implemented using MATLAB because of its popularity among acousticians. This allows users without a strong programming background to easily study the written code and modify it according to their needs. The data flow between the different components of the platform is summarized in Figure 2.

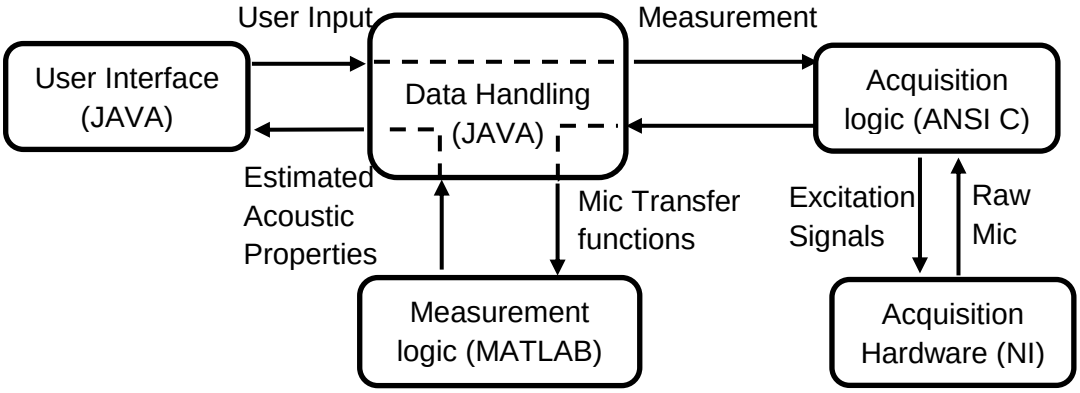

Figure 2: Data flow between different modules of the suggested platform. The technology used to develop each module is indicated.

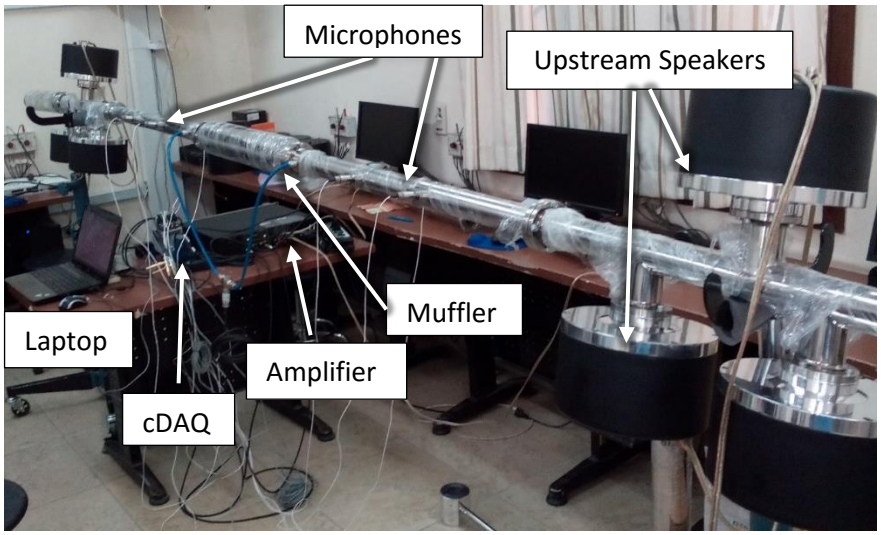

Figure 3: The two source method test setup for evaluating the acoustic properties of mufflers.

4 Acquisition algorithm

In order to perform two source measurements, the speakers could be excited using band limited white noise or stepped sine signals. When using white noise excitations, all the frequencies up to the plane wave limit of the tube are excited at once. This allows for the estimation of the acoustic properties for the excited range of frequencies in one step. The stepped sine measurements, on the other hand, are performed by exciting each speaker with a pure sinusoidal tone and estimating the transfer functions of the microphones for the excited frequency only. The target frequency range is covered by changing the excitation frequency in steps. When compared to white noise measurements, stepped sine measurements usually take significantly more time to cover the same frequency range; however, they offer the advantage of concentrating the power of the speakers around a single frequency. This means that the signal to noise ratio of the measurements is usually higher. This is important when measuring the

acoustical properties of mufflers in the presence of flow where the noise level is significantly higher than the no flow case, because of the presence of flow generated noise.

4.1 Conventional stepped sine acquisition algorithm

An example of a conventional acquisition algorithm for stepped sine excitations is that implemented in the commercial software SIDLAB [4]. The algorithm operated according to the flowchart show in Figure 4a. It receives the user requested frequency range (f_{min}: f_{max}), frequency step (f_{step}), number of averages (N_{av}) and the measurement direction (Upstream or Downstream). Then it estimates the sampling frequency (f_s) of the input and output channels so that they are greater than the Nyquist frequency for the entire frequency range.

$$f_s \geq 2\, f_{max} \tag{1}$$

The number of data samples to collect before converting to the frequency domain (Window size W_{sz}) is determined from the relation:

$$W_{sz} = 2 \times \frac{f_s}{f_{step}} \tag{2}$$

This ensures that the frequency spacing in the frequency domain Δf is half f_{step} since:

$$\Delta f = \frac{f_s}{W_{sz}} \tag{3}$$

It sends a sinusoidal wave pattern with frequency f_{min} to the output channel. It then waits for a predetermined time for the system to settle (Settling time (t_s)), before acquiring the microphone signals. When a data window is acquired, it is then windowed using a Hanning window, and converted to the frequency domain using Fast Fourier Transform (FFT). The frequency response of each mic for the excitation frequency is saved and the rest of the frequency spectrum is discarded. This step is repeated until the desired number of averages is acquired. The averaged cross spectrum between each microphone and the feedback signal from the speaker is calculated from:

$$\bar{S}_{fn} = \overline{conj(R_f) \times R_n} \tag{4}$$

where R_f denotes to the single sided frequency response of the feedback signal at the current frequency and R_n is that of the microphone number (n) and the ($\bar{}$) sign indicates the average. The coherence between the excitation and each mic (n) can be estimated from the relation:

$$Coherence_n = \frac{|\bar{S}_{fn}|}{|\bar{S}_{ff}| \times |\bar{S}_{nn}|} \tag{5}$$

The transfer functions between each mic and the first mic (H_{1n}) are simply:

$$H_{1n} = \frac{\bar{S}_{fn}}{\bar{S}_{f1}} \tag{6}$$

The excitation is then stopped, the target excitation frequency is incremented by f_{step} and the previous steps are repeated for each frequency step up to f_{max}. The time required to measure the transfer functions for a given frequency range for both the upstream and downstream directions, ignoring the processing time of the data is given by:

$$t_a = \left(N_{av} \times W_{sz} \times \frac{1}{f_s} + t_s\right) \times 2 \times N_{steps} = \left(N_{av} \times \frac{2}{f_{step}} + t_s\right) \times 2 \times N_{steps} \tag{7}$$

4.2 Optimized stepped sine acquisition algorithm

Two main aspects of the conventional algorithm could be modified to reduce the acquisition time. The sampling frequency and the window size in the original algorithm depend on the maximum frequency and the frequency step defined by the user. But, since each frequency is estimated separately from the entire frequency range, the window size and sampling frequency should only depend on the current excited frequency. The other aspect is that the excitation is stopped and restarted after each frequency step. This suggests that greater settling times are expected for the speakers and the excitations to settle.

The first aspect is approached as follows; the window size will be fixed to a constant number in the order of 2^n, this would simplify the memory management process and allow for faster FFT conversions. The sampling frequency on the other hand will be chosen so that a certain number of periodic cycles (N_{cycles}) for the current frequency is included in each data window:

$$f_s = \frac{W_{sz}}{N_{cycles}} \times f \tag{8}$$

where (f) is the current excitation frequency. It should be noted that the sampling frequency will never be less than the Nyquist frequency as long as $W_{sz} \geq 2N_{cycles}$.

To avoid stopping and restarting the excitation between frequency steps. The output channel is configured for continuous excitation and only the frequency of the excitation is changed, this allows for smoother transition between frequency steps and a reduced settling time. As a result, the acquisition time is changed to:

$$t_a = \sum_{f=f_{min}}^{f_{max}} \left(N_{av} \times \frac{N_{cycles}}{f} + t_s\right) \times 2 \tag{9}$$

This indicates that the acquisition time is inversely proportional to the measured frequency, and since the measured frequency will be usually greater than the frequency step, it is expected that acquisition time will be reduced by a factor, which depends on the measurement frequency range.

4.3 Simultaneous stepped sine excitation

It is noted in the previous algorithms that only one excitation source is active at a time. This means that while measuring in the upstream direction, the downstream speaker is not being used. This is a waste of resources especially that only one frequency is targeted upstream

frequency and the rest of the frequency spectrum is not used. To better utilize the measurement resources, the two speakers could be used to excite the system at the same time but with a different frequency. The frequency of each speaker should not be too close that they interfere with each other and not too far that they require, different sampling rates or window sizes.

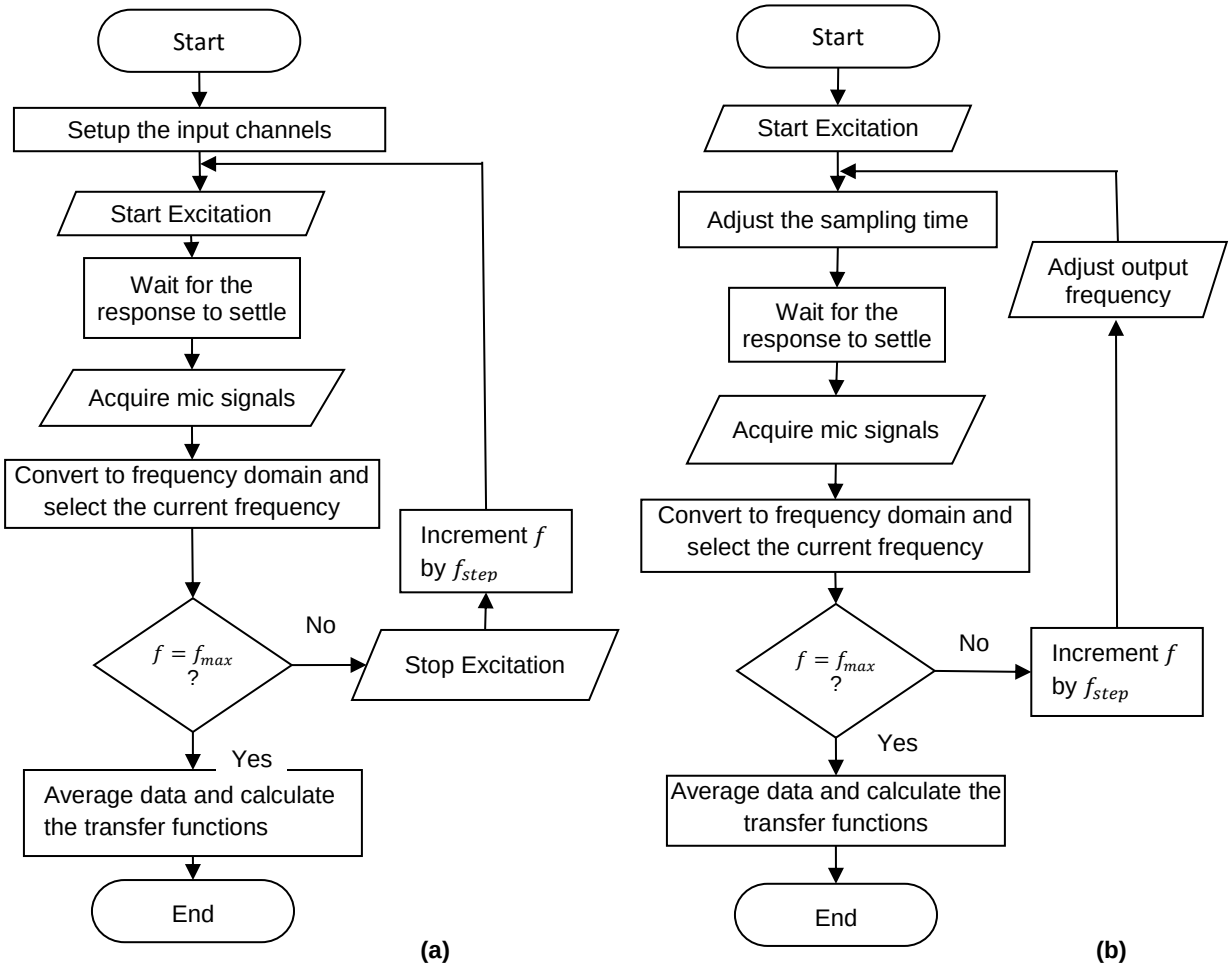

Figure 4: A flowchart of the (a) Conventional acquisition algorithm and (b) Optimized acquisition algorithm with single excitation implemented in the new platform

The new suggested algorithm would start by exciting the upstream speaker and setting the acquisition sampling time to match the upstream excitation. The acquisition algorithm would proceed as the previous algorithm acquiring the transfer functions and increasing the frequency by the required frequency step. After a predetermined frequency steps have been acquired the downstream excitation will start from the minimum frequency and simultaneous acquisition of the upstream and downstream responses will be recorded for the two excited frequencies. The two sources will be stepped simultaneously until the maximum frequency is reached by the upstream excitation. Then it will be stopped and only the downstream excitation and acquisition will continue till it also reaches the maximum frequency.

The minimum frequency spacing between the upstream and downstream excitations (Δf_{min}) will depend on the type of the windowing function. It will determine the minimum amount of frequency bins (N_{bins}) required before frequency leakage due to windowing causes errors in the measurements. The minimum frequency spacing is then given by:

$$\Delta f_{min} = N_{bin} \times \Delta f = \frac{N_{bins}}{N_{cycles}} f \tag{10}$$

When using a Hanning window N_{bins} could be as low as 2 without large leakage [8]. The maximum frequency spacing on the other hand is limited by frequency resolution of acquired data thus:

$$\Delta f_{max} = f_{upstream} - \Delta f = \frac{N_{cycles} - 1}{N_{cycles}} f_{upstream} \tag{11}$$

The absolute maximum frequency spacing for the entire frequency range is when $f = f_{min}$. It should be noted that when the frequency spacing is Δf_{max} as calculated from equation (11), the number of periodic cycles acquired for the smaller frequency is only one periodic cycle. So for practical purposes, it is recommended to keep the maximum frequency spacing below $\Delta f_{max}/N_{cycles_{\min}}$, where $N_{cycles_{min}}$ is the minimum acceptable number of cycles per FFT window. If f_{step} is larger than the maximum allowable frequency spacing, then the algorithm will simply fall back to exciting upstream and downstream separately.

5 Validation of the new algorithm

A straight pipe of length 1m and a dissipative muffler were used as test cases to validate the new algorithm and compare its accuracy and performance to the conventional one. Each test case is mounted to the test rig shown in Figure 3. The rig has an internal pipe diameter of 50 mm and is fitted with 6 PCB model 378C10 ¼" IEPE microphones, which were calibrated using a B&K 4231 sound calibrator. Three SEAS W18EX001 100W low frequency speakers were used at each side to provide acoustic excitations. The speakers are driven by a 2-channel 390W Yamaha P3500S power amplifier. The outputs of the microphones and the inputs of the amplifier are connected to a NI cDAQ-9174 data acquisition system with 2x 4 channel 9234 input modules and a 4 channel 9263 output module. The cDAQ is driven via USB by PC which hosts the new acquisition platform and also SIDLAB 3.3 as the conventional reference implementation. A variable speed fan is used to generate flow inside the test rig.

The experimental results were also compared to analytic two-port network models for each test case. The models were created using the SIDLAB acoustics module.

The dissipative muffler, shown in Figure 5, consists of a perforated pipe of length 500 mm, diameter 50 mm and porosity 20%. The pipe is placed inside a cylindrical chamber of diameter 99 mm. The chamber is filled with Rockwool of density 70 kg/m^3 and flow resistivity 27000 rayl/m.

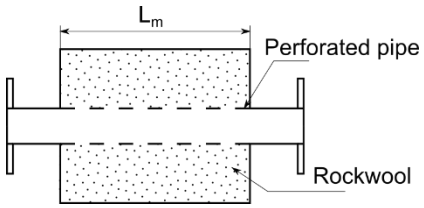

Figure 5: Schematic of the construction of the tested dissipative muffler.

6 Results

For each test case the performance of the new platform was compared to that of the reference implementation with no flow and in the presence of a flow with Mach number M=0.1. All the results were obtained for a frequency range from $f_{min} = 200Hz$ up to a $f_{max} = 1100Hz$ with a step $f_{step} = 10\,Hz$ and a settling time $t_s = 500ms$. The newly defined constants were taken to be $W_{sz} = 2048$ and $N_{cycles} = 20$. The number of averages for each measurement were 25 for the no flow case and 100 in the cases when flow is present.

6.1 Straight Pipe

Since for a straight pipe acoustic waves are almost fully transmitted without reflections, the complex transmission coefficient (S_{12} element of the scattering matrix), was used to compare the accuracy the different acquisition approaches. This is demonstrated in Figure 6, which shows excellent agreement between all the acquisition techniques and the analytic estimation. This indicates that for the tested frequency range and for the simple case of a straight pipe the accuracy of the measurements is not affected after the application of the new algorithm with single sided excitations and with simultaneous excitation being used.

6.2 Dissipative Muffler

For a more practical measurement the Transmission Loss (TL) of the dissipative muffler is used to compare the accuracy of the different acquisition approaches. For the no flow case (Figure 7a) a slight shift between the analytic estimation and measured values for all the techniques. This is acceptable given the approximate nature of the two-port analytic model. Excellent agreement between all stepped sine acquisition techniques is observed. It is noted that a few peaks are observed in the results obtained using white noise excitations which are not observed in the other approaches. This is attributed to the fact that the acoustic power is not equally distributed over the target frequency spectrum which causes the signal to noise ratio to drop at certain frequencies causing erroneous results. This was confirmed by comparing the locations of the peaks to the location were the microphone coherences drop below 0.8. A similar effect is noted in the case with flow (Figure 7b), at frequencies higher than 1000 Hz, both the conventional (SIDLAB) and the optimized acquisition algorithms produce erroneous results. This is mainly because of the presence of flow induced noise and the high transmission loss of the muffler at these frequencies, which causes the pressure amplitudes recorded by the downstream microphones to drop below the noise floor producing a noisy transmission loss at these frequencies. This problem could be mitigated by increasing the amplitude of the excitation

at the high frequency using speakers with higher operating frequency range, which were not available at the time of the measurement.

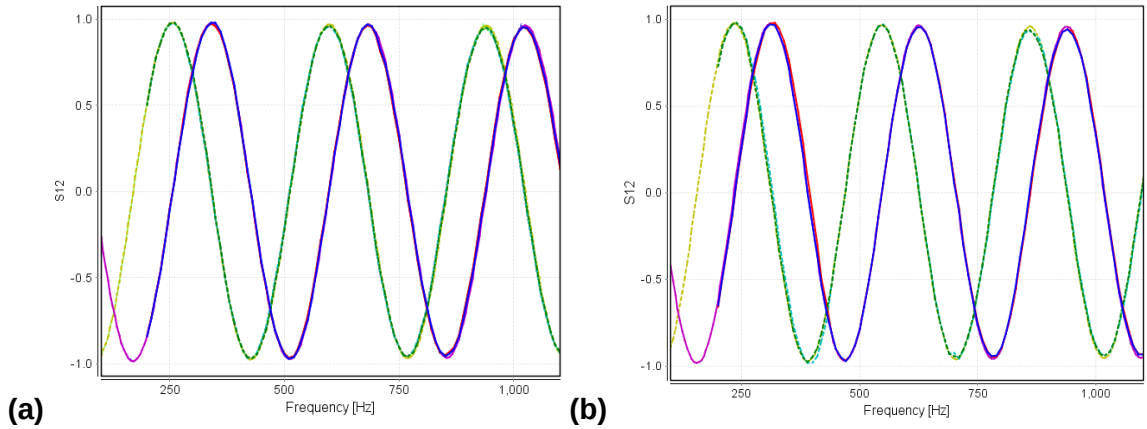

Figure 6: Real and imaginary components of S_{12} estimated using simultaneous stepped sine (Real part and Imaginary part), white noise (Re and Im), optimized single-sided stepped sine (Re and Im), conventional stepped sine excitations (Re and Im) and analytically (Re and Im) for the cases (a) M=0 and (b) M=0.1

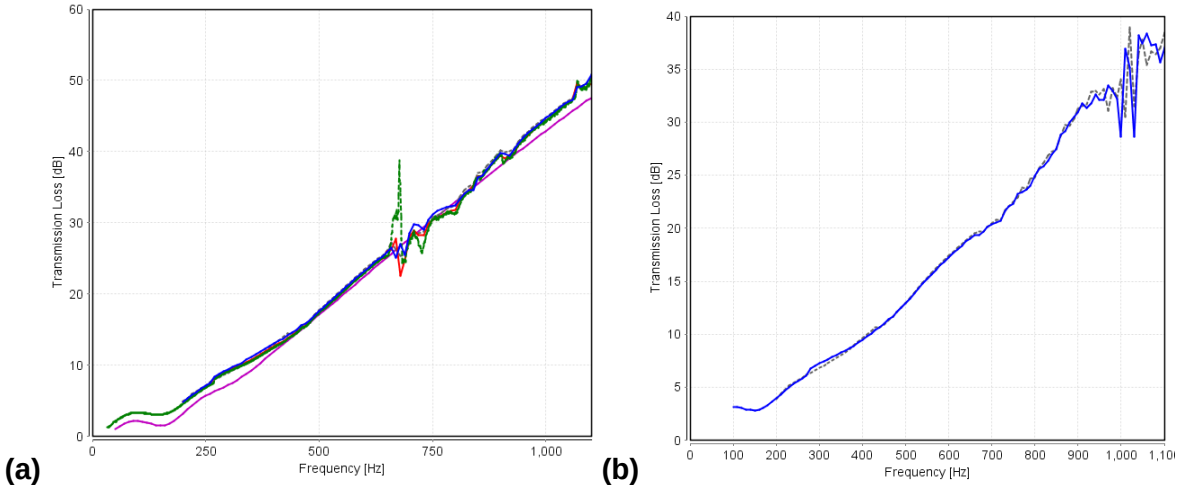

Figure 7: TL of the dissipative muffler estimated using simultaneous stepped sine, white noise, optimized single-sided stepped sine, conventional stepped sine excitations and analytically for the cases (a) M=0 and (b) M=0.1

6.3 Acquisition time

The acquisition time taken by each method was calculated as the time between sending the measurement command to the acquisition algorithm and receiving the microphones transfer functions. Table 1 shows the reduction in the acquisition time between the application of the new algorithm, while keeping the quality of the measurement as shown in Figure 6 and Figure 7. The new platform without the simultaneous excitation is 3.5 times faster for the tested frequency range. This speed up is attributed to the conventional algorithm assuming worst case scenarios that were simply not present. Another speed gain of 1.5 is achieved by the use of

simultaneous excitations making the overall achievable reduction in acquisition time by a factor of 5.3. The fact that the speed up due to the simultaneous excitation is not 2, is clear since the two speakers cannot be simultaneously excited for the entire of the frequency range.

Table 1: Comparison of the total acquisition time taken by the different stepped sine excitation algorithms within a frequency range of 200 and 1100 Hz, a step of 10Hz and 100 averages.

Acquisition algorithm	t_a (min)	Speed Gain
Conventional	32	-
Optimized	9.13	3.5
Optimized with simultaneous excitation	6	5.3

7 Conclusions

A new platform for the automation of two source acoustic measurements is introduced. It includes an optimized algorithm for performing stepped sine measurements which minimizes the time needed to acquire the necessary data without affecting the quality of the measurement. The new algorithm implements simultaneous upstream and downstream stepped sine measurements. A stepped sine measurement ranging between 200 and 1100 Hz with a 10 Hz step and 100 averages per step is finished in 6 mins by the new optimized algorithm which makes it more than 5 times faster than a conventional algorithm currently implemented in the commercial software SIDLAB 3.3, with almost no effect on the quality of the acquired results.

References

[1] Tao Z.; and Seybert A. F. A review of current techniques for measuring muffler transmission loss, *SAE Tech. Pap.*, Vol. (2003–1), 2003, pp 1653.

[2] Pedrosa A. M.; Denia F. D.; Fuenmayor F. J.; Abom M. Experimental measurement of acoustic properties by a two source method with simultaneous excitation, *Proceedings of 18th International Congress of Sound & Vibration*, 2011.

[3] Pedrosa A. M.; Denia F. D.; Besa A. J.; Fuenmayor F. J. A two source method with simultaneous excitation for the acoustic characterization of exhaust systems with mean flow, *Proceedings of INTER-NOISE and NOISE-CON Congress*, Vol (247), 2013, pp 3783–3790.

[4] Elnady T. A platform to measure transmission loss of mufflers with flow, *Proceedings of Inter-Noise and Noise-Con Congress and Conference Proceedings*, 2007, pp. 3418–3427.

[5] Abom M. Measurement of the scattering-matrix of acoustical two-ports, *Mech. Syst. Signal Process.*, Vol (5), 1991, pp 89–104.

[6] ASTM E2611 – 09: *Standard Test Method for Measurement of Normal Incidence Sound Transmission of Acoustical Materials Based on the Transfer Matrix Method*, 2009.

[7] Lavrentjev J.; Abom M.; Bodén H. A measurement method for determining the source data of acoustic two-port sources, *J. Sound Vib.*, Vol 183 (3), 1995, pp. 517–531.

[8] Harris, F. On the use of windows for harmonic analysis with the discrete Fourier transform. *Proceedings IEEE*, Vol 66 (1), 1978, pp. 51–83.