

High-Frequency Time Domain Source Path Contribution: From Engine Test Bench Data to Cabin Interior Sounds

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ABSTRACT

This work presents an application of airborne source path contribution analysis with emphasis on prediction of wideband sounds inside a cabin from measurements made around a stand-alone engine. The heart of the method is a time domain source path receiver technique wherein the engine surface is modeled as a number of source points. Nearfield microphone measurements and transfer functions are used to quantify the source strengths at these points. This acoustic engine model is then used in combination with source-to-receiver transfer functions to calculate sound levels at other positions, such as at the driver's ear position. When combining all the data, the in-cabin engine sound can be synthesized even before the engine is physically installed into the vehicle.

The method has been validated using a powertrain structure artificially excited by several shakers playing band-limited noise so as to produce a complicated vibration pattern on the surface. First the excited structure is studied alone; next a vehicle cabin was lowered onto the structure without touching. As a result we can compare the combination of using only in-vehicle operating/transfer data or using powertrain only operating data and in-vehicle transfer data for synthesizing interior sounds. Very good agreement between the two procedures was obtained and comparable to the actual sound measured inside the cabin during operation.

In addition to verifying the above procedure, the same near-field microphone setup, whether powertrain alone or with cab on top, can be used to assess the radiated sound power from the vibrating structure. The procedure is outlined and the obtained sound power spectra are validated against a standardized hemisphere sound power measurement showing very good agreement in general.

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INTRODUCTION

Modern vehicles continue to become more refined, with a higher customer expectation of NVH refinement. Airborne sound radiated from the surface of the powertrain is of course a significant factor in overall perceived NVH refinement. Some powertrains, including diesel engines and direct-injection gasoline engines, tend to have a higher level of high-frequency impulsive sound content. Other alternative powertrains, e.g. hybrid, electric, fuel-cell, can have sounds including transient, high frequency sounds which need to be controlled so as to not affect the overall perceived refinement of the vehicle.

Source path contribution analysis techniques have been used for many years to quantify the source levels, sound and vibration transmission paths and contributions from the various sources and paths to receivers in vehicles for both airborne and structure-borne cases. These analyses have typically been performed in the frequency domain, and for airborne cases, have typically not been performed at high frequency.

Here, a time-domain airborne source-substitution source path contribution technique has been used to quantify the sound radiated from the surface of a powertrain, and calculate in-vehicle levels from the measurements made around the powertrain both for the in-cab condition as well as the powertrain-only condition.

The goals of this measurement and analysis activity were to 1) validate the airborne time-domain source path contribution calculations to 10kHz or above, 2) calculate in-cabin levels from powertrain-only measurements, and 3) calculate sound power levels of the powertrain from free-field powertrain only condition and in-vehicle condition.

One of the benefits of this approach is that the results, both the source levels and contributions from each source to the receivers, are in the time domain, which preserves the transient character of the sound. This allows for both listening to the contributions and the overall calculated interior sound level, as well as additional post-processing of the transient signals.

Another benefit of the source-substitution technique is that the source levels can be measured in any environment, and then used with any appropriate set of transfer functions to calculate interior sound levels inside the vehicle. This means that source levels can be measured in an engine test cell, and used to calculate what the engine will sound like in a vehicle. Or, the source levels can be measured in one vehicle, and used to calculate what that engine will sound like in a completely different vehicle.

Yet another benefit of this approach is the ability to estimate sound power levels from the calculated source levels. It is often useful to understand the overall sound power level of the powertrain. This typically requires either the powertrain placed in a sufficiently large hemi-anechoic environment, or a sound intensity scanning method to measure sound power. Both of these methods have their

obvious disadvantages. Using the calculated source levels from the source-substitution method, sound power can be estimated with a reasonable level of accuracy without the limitations of the traditional sound power measurement techniques, and can be calculated from measurements made in any test cell or event with the powertrain installed in a vehicle.

TIME-DOMAIN SPC APPROACH

Experimental modeling methods based on acoustic transmission path analysis, or SPC (source path contribution) are typically employed when a certain number of source positions are defined close to the true acoustic sources [1, 2]. As such each relevant noise component of the vehicle, e.g. engine, intake and tire is modeled via a set of assumed point sources covering a component and being able to represent the radiated sound field. Next step is to quantify the source strength of each modeled point sources and finally we can predict the noise at a certain position from one component only. The modeling principle is sometimes termed *source substitution method*.

As an example consider the complex noise source encapsulated by some structure in [Figure 1](#) resembling the situation of an engine inside an engine compartment. The noise generated by the vibrating surface is modeled using a set of distributed point sources and each source's operational strength is estimated using a set of indicator microphones preferably located close to the point sources. To accomplish this acoustic transfer functions are measured between all point sources and all indicator microphone positions to allow full matrix coupling between all sources and indicators. A calibrated volume velocity source is used for this task. If the model is used to predict the operational response at a receiver location we can measure this acoustic transfer function at the same time. Then the actual noise source is turned on and operational data is recorded for all the indicator microphones, and if possible also at a desired receiver microphone as illustrated. Since the operational data and transfer function data are both acquired within the same environment they fit together and we can firstly quantify the operational source strength for each point source given the matrix of transfer functions and the indicator operational data. Here the matrix of transfer functions is usually provided as a set of Frequency Response Function's (FRF's) measured by the volume velocity source as sound pressure over volume velocity (p/Q) FRF's.

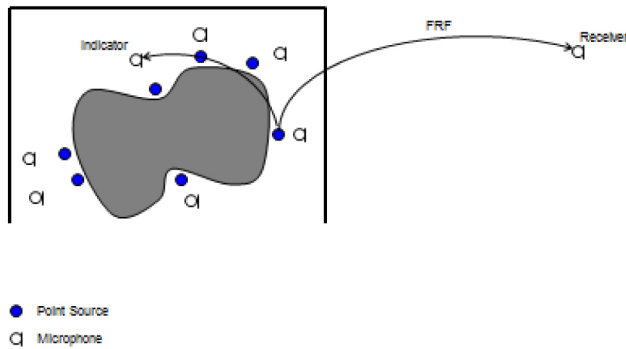


Figure 1. Source substitution method applied to a complex vibrating geometry encapsulated by another structure

Since all indicator microphones receive contributions from all noise components to a greater or lesser degree, we have to set up the relation between source strengths and indicator sound pressures as a matrix formulation:

$$[q_{src}] = [H]^{-1}[p_{ind}] \quad (1)$$

where H is the matrix of source-to-indicator FRFs, q is the vector of source strength and p is the vector of measured indicator sound pressures. In principle we can then invert H and multiply with p to obtain a set of source strengths that fit our measured indicator pressures via our measured source-to-indicator FRFs. In many situations H will have a rectangular form since more indicator microphones than source positions are chosen so in that case we will solve [equation \(1\)](#) in a least squares sense. This equation is usually solved in the frequency domain, however since our operating data exists as time recordings of fast run-up's, we prefer to solve the matrix problem directly in the time-domain.

One problem common for all matrix solvers has to be dealt with, the possibility of having an ill-conditioned and thus unstable FRF matrix. Stability in terms of regularization can be introduced if some small terms (viewed as spurious data or noise) are removed from the FRF matrix, however if too much information is removed the estimated source strengths become inaccurate. A singular value decomposition (SVD) routine is run for every frequency line of the FRF matrix, the set of singular values will be grouped into two groups: 1) retained set and 2) discarded set of singular values. The retained set will consist of the largest singular values since they account for most of the information in the FRF matrix. The SVD routine returns singular values arranged from highest to lowest, and a threshold dividing the set into the two groups is chosen. This chosen threshold is relative to the largest singular value for every frequency line, that is a threshold of 1/10 (or 20dB) means all singular values from the largest down to 0.1 times the largest will be put into the group of retained singular values, the rest below will be discarded. As such the group of discarded singular values are assumed to represent noise and other errors in the FRF

matrix. A default threshold value is normally used for every frequency line however one can fine tune this value if pre-knowledge about the true error levels of the FRF matrix and also the indicator operating data is available.

In the following approach we will solve for the estimated operational source strength's using a time-domain procedure. The inverse of the FRF matrix is turned into a matrix of inverse filters which are used to perform multi-channel inverse filtering on the operational indicator time data recorded during the operating noise source [3]. We can write this operation as:

$$[q_{src}(t)] = [h(t)] * [p_{ind}(t)] \quad (2)$$

where $[h]$ represents the matrix of finite impulse response (FIR) inverse filters based on the inverse of measured FRF's. Here care must be taken when inverting the FRF matrix since it is prone to ill-conditioning and regularization techniques would be necessary. The convolution operator '*' is introduced at this stage.

The set of operational source strengths for the complex noise source constitutes a source descriptor. Given another set of transfer functions from the same point source positions to a receiver microphone position we can predict the noise generated at the receiver from the complex noise source by adding up the contributions from each source position:

$$p_{rec}(t) = \sum g(t) * q_{src}(t) \quad (3)$$

where g is now a forward FIR filter between source and receiver. The delays introduced by the filters in h and g can be removed so that the resulting predicted time data will be synchronized with the original measured dataset. If we had measured the noise at the receiver during source operation we can in fact validate the result from the prediction and assess the quality of the noise source modeling.

Measurements

A detailed measurement campaign was carried out where the powertrain structure was first positioned in a hemi-anechoic chamber above the hard ground. An initial set of indicator microphones was placed around the complete structure, 43 microphones in total for picking up the noise generated by the shaker excitation. A compromise between having the exact same distance between the structure and each microphone and using simple line-arrays of microphones for easy setup was made. A HATS (Head and Torso Simulator) binaural receiver was considered for free-field validation of the acoustic model obtained from the SPC measurements.