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MODELLING IN-VEHICLE ENGINE NOISE (LISTEN TO THE NOISE PATHS)

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Transfer Path Analysis is accepted as a tool to model or trouble shoot the lower engine orders for in-vehicle noise. The contribution of each force input that is coherent with the engine can be evaluated both for amplitude as phase. This allows to rank the individual paths and allows to identify paths that interact (cancel effects) . The ability to listen to these contributions also called partial pressures, is a need for the sound quality evaluation of engine noise, and the next logical step in Transfer Path Analysis. Steady-state applications of Time Domain Transfer Path Analysis have been reported, but primarily the study of pseudo-stationary sounds such as engine run-up's was desired. In this document, different methods will be discussed that could enable this type of work. The new developed Synthesized Order Substitution method is presented. This method enables an engine sound engineer to listen to the sound due to a single or a set of noise transfer paths, for a reasonable computation effort.

INTRODUCTION

The study of transient or pseudo-stationary sounds such as engine run-up's needs tools to better understand the sound quality message or sound quality perception of the dominant lower engine orders. The purpose of these tools is to enable a sound quality engineer to perform "what if ..." scenarios with the order content of the time domain binaural signal. These "what if ..." scenarios require that the original order content can be modified to whatever order content provided by that engineer. The most suitable format for these new orders is an amplitude / phase curve as a function of engine speed. Tools like this would be most welcome in applications such as Transfer Path Analysis. The ability to listen to the individual paths has been a high priority feature for many years for automotive sound quality engineers.

A number of attempts have been made to come to Time Domain Transfer Path Analysis. Up to today serious problems were encountered in getting a principle to work. In this publication an overview and evaluation of three methods, originally considered to have potential for this Time Domain TPA application will be presented. These methods are based on: FIR and IIR filter techniques; long FFT's; or a new method using synthesized order substitution.

TIME DOMAIN FIR OR IIR FILTERING

The principles and formulas of Transfer Path Analysis are rather simple, and can be expressed in the time domain:

$$y(t) = \sum_{i=1}^{n} \int_{-\infty}^{\infty} h_i(t-\tau) \cdot f_i(\tau) \cdot d\tau$$

For the determination of forces only the mount stiffness method is supported:

$$f(t) = \int_{-\infty}^{\infty} k_i(t-\tau) \cdot (x_{b,i}(\tau) - x_{e,i}(\tau)) \cdot d\tau$$

The use of time domain filter techniques for this application does require a non-negligible computation effort. This computation cost would be no problem if this method would be suitable for the lower order or lower frequency applications. The real-life noise transfer functions "H(f)" have a very complex amplitude and phase behavior. In the frame of the European BRITE Project SOQCRATES, this matter was investigated. Despite the use of high order FIR and IIR filters, these were not able to reproduce this complex shape. The sound of the calculated signals "y(t)" is not close enough to the real signals for sound quality applications. Add to this the fact that mount stiffness spectra are not available for several force paths, and that the mount stiffness method is not applicable for some force paths such as direct inputs.

This method was not seen as suited for lower order Time Domain TPA.

LONG FFT TRANSFORM FOR PSEUDO-STATIONARY SIGNALS

Next to linearity, one of the properties of FFT is that there is an equivalence between certain operations in the frequency and certain operations in the time domain. The multiplication in the frequency domain is a lot more straightforward than the equivalent convolution in the time domain. Therefor it seemed reasonable to test this out for the basic frequency domain formulation of TPA:

$$Y(f)=\sum_{i=1}^{n}H_i(f)\cdot F_i(f)$$
$$F_i(f)=K_i(f)\cdot\frac{X_{b,i}(f)-X_{e,i}(f)}{-(2\cdot\pi\cdot f)^2}$$

The main problem of the use of FFT is the fact that the method assumes that the signal is periodic, or that the frequencies part of the signal are limited to the ones of the spectral lines of the FFT. This is by definition not the case for a pseudo-stationary test condition such as an engine run-up.

The use of a long FFT has the advantage that the frequency resolution becomes quite small. For an engine run-up of 20seconds, the frequency resolution is 0.05Hz. A disadvantage is that the total segment size that is being investigated has to be a power of two. The computation times that are needed are acceptable and considerably lower than the ones for time domain filters.

The following test was executed:

1. a sine sweep of 10 seconds from 20 to 200Hz was generated in the time domain;
2. a noise transfer function "H(f)" with a typical complex shape was selected;
3. a long FFT was applied on the sweep;
4. a complex multiply of sweep spectrum with the noise transfer function spectrum was performed;
5. an inverse FFT on the multiplied spectrum "Y(f)" resulted in a full audio time trace.

While listening to "Y(f)", amplitude oscillations could be heard especially just before important increases in amplitude or after important decreases in amplitude. These amplitude oscillations were visually noticeable in the time domain. Once a listener had noticed these artifacts, their influence on the perception of the signal became too important.

Although the long FFT method seemed practical for a time domain TPA development, it could not be used for pseudo-stationary applications for the disturbing artifacts that are created in the numerical process.

SYNTHESIZED ORDER SUBSTITUTION

The above mentioned methods all suffer from the fact that the impulse response or FRF of the different noise transfer paths are needed. These functions typically are only accurate up to about 500Hz (exceptions up to 1500Hz have been reported). For a majority of cases, a good perception of the in-vehicle sound requires not only the lower orders, but also the high frequency noise that is wind, road and engine induced, and the other non-harmonic components in the lower frequencies. Therefor, a method was investigated that would try not to disturb the available sound in the process of substituting the existing order content by the desired order function. This with a focus on keeping the phase as close as possible to the desired amplitude and phase. This method had also to be compatible with the partial pressure results of a frequency domain based Transfer Path Analysis tool. In this way, also the more advanced inverse force identification methods can be used in the global time domain TPA process.

The synthesized order substitution procedure has the following steps:

1. generate a target order function with exact absolute phase;

2. generate a time sweep for that desired target order function;
3. eliminate the order content of the target order from the original full audio time signal using double precision ultra-sharp tracking order filters;
4. insert the sweep signal in the filtered signal, and listen.

A series of difficulties needed to be tackled to get this 'simple' procedure operational. Most difficulties were encountered in getting the absolute phase for order spectra obtained from fixed sampled data.

Absolute phase determination

The outputs of a TPA model are the partial pressure spectra. These spectra are complex valued spectra over a given RPM-range. The phase of these spectra is relative to a reference defined in the acquisition of the operational data. Two methods to obtain a reference phase are commonly used: either a trigger channel is selected, or, most commonly, a reference channel is selected. It is important that the trigger or reference channel do contain clearly the half order. Next to that, either fixed sampling acquisition techniques or synchronous sampling techniques can be applied. For an accurate phase on the combinations fixed-sampling with reference channel and synchronous-sampling with trigger yield proper results. In our study the focus was on full audio fixed sampling recordings, thus using reference channels.
The phase of a spectrum obtained from FFT on an pseudo-stationary engine run-up is not reliable. However, the phase difference when using reference channels, is very accurate. This is probably due to the fact that the FFT distorts the phase in a similar way for both target as reference channel. The equation describing the processing involved when using a reference channel 'B' for a signal 'A', is given below:

$$A \cdot e^{jr} = A \cdot e^{ja} \cdot \frac{B}{B} \cdot e^{-jb} = A \cdot e^{j(a-b)}$$

This leaves us with the problem of obtaining the absolute phase 'b' of the reference signal in order to be able to calculate the absolute phase 'a' needed.
The absolute phase of the reference signal is obtained using angular domain re-sampling with the trigger channel as rpm/angle reference. Once the reference and trigger signals are converted to the angular domain, a triggered measurement will yield absolute phase for the reference signal with respect to the trigger moments in the trigger channel. Since accurate phase is the desired parameter, an important over-sampling factor needs to be used to be sufficiently below the low-pass tracking filter used in that process.
An alternative method is to convert the target signal itself to the angular domain and calculate the absolute phase 'a' directly. This method will be used in our feasibility study presented below, but this direct method is for practical TPA-work not optimal since phase components from non-harmonic components typically present in an acoustical signal will influence the absolute phase results.

Order sweep generation

Within the SOQCRATES project an advanced signal generator was developed. This signal generator allows to generate sine sweeps with predefined order, amplitude and phase content. This tool is a key element in the total procedure. The conversion of the unequally spaced order traces with RPM as abscissa with a typical resolution of about 20 to 60 RPM, to equally spaced order-, amplitude- and phase-traces in the time domain sampled at 48kHz, was the next concern.

Especially the calculation of the phase trace is the most cumbersome task. Small judgment errors in the phase processing result in audible artifacts. Currently, a satisfactory algorithm is implemented, however, it is in this area that additional effort will be spent to come to even better performance.

Ultra-sharp time domain order filtering

Through the use of double precision tracking filters, the existing order content of a measured target signal is removed. The performance of the filters is outstanding: order width 0.01 order combined with an attenuation of -40dB. The phase distortion of these filters outside the filtered order range is below 10°. In consequence of the small filter bandwidth, the accuracy of the RPM estimate has become the sensitive parameter in the order filter process. Therefor, the order filter algorithm uses an interpolated RPM estimate for the filter coefficients valid at each time sample. Ultra-sharp filters are typically not able to follow fast changes in amplitude of the component that they are filtering. For our application this is mainly a positive aspect, since the side-bands resulting from modulation on the orders, are not affected in this filtering process. This is very important feature when trying to reproduce Roughness and Fluctuation Strength adequately. Next, when filtering the side-bands, the modulation character can be removed and eventually replaced.

Insertion of the synthesized sweep

Finally, the synthesized sweep has to be inserted in the filtered target signal. Since phase was the key element from the start, this operation has to be done accurately. The absolute phase from the initial target orders was calculated relative to a trigger signal. This trigger signal was used to obtain the absolute phase of the reference signal in the angular domain, and quite often also to calculate the RPM. The synthesized signal also starts at such a trigger point, and therefor the insertion has to be exactly at the trigger point that corresponds to the start RPM of the filtered target signal.

Initial test results

The synthesized order substitution method has tested with a number of engine run-up's that were available. The purpose of the tests was to check the quality of the procedure through a closed loop test. The different steps in the tests are given below:

1. From the engine run-up, orders were determined using FFT-based order analysis for fixed sampled data. These orders were spectra as a function of RPM. The RPM resolution has to be sufficiently small in order to minimize the phase jumps between consecutive RPM points within an order. This resolution is depending on the order and even more on the signal itself.
2. The RPM was determined from the tacho trace, and an RPM-value was estimated for each time sample part of the engine run-up.
3. The trigger trace (typically 2, 2.5 or 3 pulses/rev) was converted to a trigger signal (1 pulse for 2 rev's) using a pulse divider algorithm.
4. One of the orders was selected and filtered from the target engine run-up time signal.
5. The absolute phase of the selected order was calculated from the target run-up signal itself and the trigger trace, after the conversion of both traces to the angular domain.

6. The phase correction spectrum 'a' was copied into the order versus RPM trace.
7. An order sweep was generated from the corrected order versus RPM trace.
8. The synthesized sweep was inserted at the appropriate location in the filtered target signal.
9. The result was compared to the original target time signal.

Listening tests learned that the overall sound perception was almost identical. The roughness perception which was dominant in some RPM-ranges did not change. However, small differences in level of the order could be heard. A comparison of the spectra of the original run-up's and the processed run's learned that indeed some amplitude and phase errors can be measured, but that for the amplitude these errors are quite small, and for the phase that the trend is almost always there, but that a phase drift is present. It is this phase drift that needs further investigation.

Conclusions

The concept of replacing the existing order content of a transient, broad-band signal by a set of synthesized orders for which the original order content was removed through the use of ultra-sharp time domain filters earlier, has proven to be a working method for sound quality engineering work.

Time domain filter techniques and long FFT techniques both suffer from either amplitude and phase reproduction quality, or audible undesired amplitude oscillations.

The new synthesized order substitution method still needs further investigation on phase behavior for longer time series, and exposure to other similar applications.

The fact that the method - through the use of double precision tracking filters - hardly affects the original signal content in time or frequency domain, makes it an ideal tool for Time Domain Transfer Path Analysis.

Time Domain Transfer Path Analysis will enable the automotive sound quality engineer to better understand the mechanisms and causes of perception of the dominant lower order components within the total engine noise, even for pseudo-stationary test conditions.