ANALYSIS, RESYNTHESIS, AND INTERPOLATION OF THE INTERIOR NOISE OF A CAR

Christian Feldbauer, Robert Höldrich

Institute of Electronic Music and Acoustics
University of Music and Dramatic Arts, Graz, Austria
feldbauer@iem.kug.ac.at, robert.hoeldrich@kug.ac.at
iem.kug.ac.at/~feldbauer/, www.kug.ac.at/iem/info/personal/rh.htm

ABSTRACT

The aim of this work was to develop a realtime simulator for vehicle interior noise in varying load situations. Using this simulator, laboratory hearing tests which are common in the automotive industry may be executed more realistically than by using recorded sounds without any interaction possibility. A simplified model for the load situation of a car is presented. Using this model, the interpolation between the prerecorded signals is performed to obtain signals for other load situations. Furthermore, an analysis/resynthesis system for the interior noise of cars is introduced which only needs a sparse set of parameters.

1. A SIMPLE MODEL FOR THE LOAD SITUATION

As a starting point for the simulator, we use speeding up and slowing down recordings for various accelerator pedal positions. To simplify the model, a constant gear and constant road conditions are assumed. In figure 1 this model is illustrated. The parameters

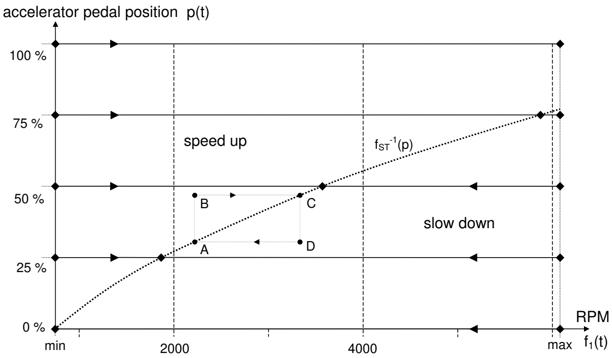

Figure 1: *The Load Situation Model.*

RPM (or $f_1(t)$) and position of the accelerator pedal $p(t)$ are used to describe the load situation of a vehicle. The so-called stationary curve (dotted line) is the border between acceleration (upper area) and deceleration (lower area). A recording of a speeding up or slowing down at a constant acceleration pedal position can be drawn in as a horizontal line which normally ends at the stationary curve. The aim is to calculate sounds for arbitrary pedal positions and speeds using the recorded sounds.

1.1. The Load Interpolation

It is necessary to interpolate between given sounds as illustrated in figure 2. We use linear interpolation between signal frames m_i

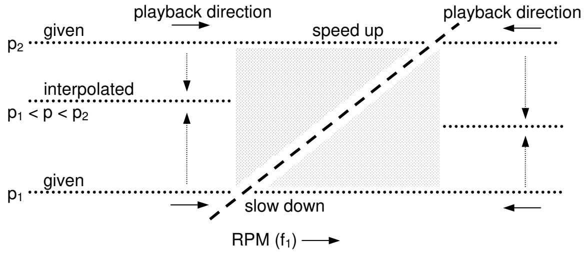

Figure 2: *The Load Interpolation.*

with equal RPM values for all parameters of the signal model:

$$w_p = w_{p_1}(m_1) + \Big(w_{p_2}(m_2) - w_{p_1}(m_1)\Big)\frac{p - p_1}{p_2 - p_1} \quad (1)$$

where w represents any parameter of the chosen signal model (see section 3.1).

The inverse functions of the RPM trajectories are essential to find the frames with the correct RPM values of the recorded sounds: $m_1 = m_{p_1}(f_1)$ und $m_2 = m_{p_2}(f_1)$

The shaded area in figure 2 shows where this kind of interpolation can not be used, because only one sound provides frames for the corresponding RPM values. A simple solution to this problem is to use the frames of the stationary point of the "shorter" signal. It's important to mention that speed-up and slow-down signals should never be mixed.

2. THE SIMULATOR

Figure 3 shows the algorithm of the realtime simulator. At first, one has to choose an RPM value f_1 initializing an endless loop. One cycle takes the time of the frame interval T_F which is normally chosen between 20 and 50 ms. Starting a new cycle, the current accelerator pedal position p has to be determined. So one can find the two corresponding soundfiles which are used for the load interpolation between p_1 and p_2. Utilizing the inverse of RPM as a function of time, we obtain the appropriate frames m_1 and m_2. The load interpolation of section 1.1 can be performed to obtain a set of new signal parameters and the sound resynthesis (see section 3.3) can be carried out. The last step is the calculation of the new RPM for the next cycle:

$$f_1 = f_1^{p_1}(m_1{+}1) + \Big(f_1^{p_2}(m_2{+}1) - f_1^{p_1}(m_1{+}1)\Big)\frac{p - p_1}{p_2 - p_1} \quad (2)$$

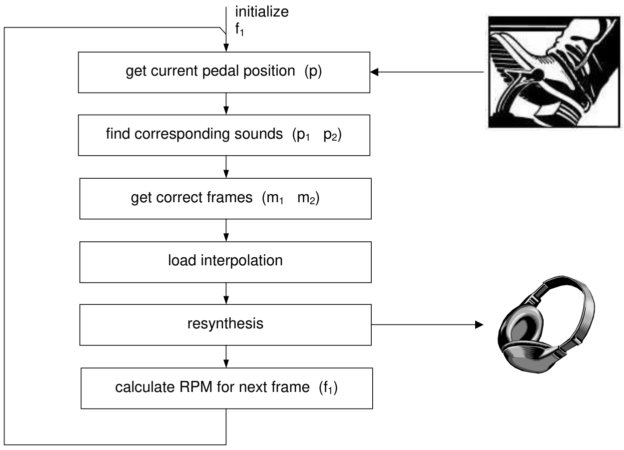

Figure 3: *The Simulator Algorithm.*

3. THE ANALYSIS/RESYNTHESIS SYSTEM

We examined various signal models and methods for the decomposition of deterministic and stochastic components, e.g. SMS [1] or QUASAR [2]. For the analysis of the deterministic component, Serra in SMS suggested to perform a peak detection followed by a peak continuation. This does not work for the interior sound of cars because the small signal-to-noise ratio results in too many candidates in the peak detection process. Thus the peak continuation can't find satisfying trajectories automatically.

3.1. Signal Model

We consider a real valued signal $x(n)$ as a sum of a deterministic $det(n)$ and a stochastic component $res(n)$.

$$x(n) = det(n) + res(n) \tag{3}$$

The main part of the deterministic component is caused by the engine, so $det(n)$ can be regarded as a harmonic complex. The fundamental frequency $f_1(n)$ and the instantaneous amplitudes of the K partials $A_k(n)$ are sufficient for the description.

$$det(n) = \sum_{k=1}^{K} A_k(n) \cos\left(2\pi k \sum_{u=0}^{n-1} f_1(u)T + \varphi_k\right) \tag{4}$$

T is the sampling interval and φ_k the phase offset of the k^{th} sinusoid. As a model for the stochastic component, we interpret $res(n)$ as time-variant coloured noise. In [3], various methods for the modelling of the stochastic signal component are mentioned.

3.2. The Analysis

The analysis is performed offline. The first step is the estimation of the fundamental frequency $f_1(n)$. We either use an RPM reference (recorded ignition signal) [4] or an STFT followed by a peak detection [1] and further processing. Converting RPM to frequency one has to take into account the ratio of explosions per round, e.g. a 4-stroke engine with four cylinders produces two explosions per round and a full period (one cylinder) takes two rounds.

Figure 4 shows the block diagram of the analysis task. The amplitudes of the partials can be extracted by heterodyne filters if an estimate of the fundamental frequency is given. A heterodyne filter is a demodulation device. The time-varying sinusoidal component is shifted to 0 Hz by the ring modulation with the complex conjugated carrier $e^{-j\varphi_k(n)}$. A lowpass filter $h_{LP}(n)$ eliminates the off band frequencies and the slowly varying amplitude $\hat{A}_k(n)$ remains.

$$\hat{A}_k(n) = \left(x_{dec}(n)\, e^{-j\varphi_k(n)}\right) \otimes h_{LP}(n) \tag{5}$$

The result $\hat{A}_k(n)$ is complex valued, it includes the phase difference to the instantaneous carrier phase $\varphi_k(n)$ which is calculated by using the estimate of the fundamental frequency: $\varphi_k(n) = 2\pi k \sum_{u=0}^{n-1} \hat{f}_1(u)T_{dec}$. T_{dec} is the sampling interval of the decimated signal $x_{dec}(n) = \mathrm{decimate}(x(n))$. This decimation can be performed, because the partials of the interior noise of a car are usually masked by noise at frequencies over 2 kHz. The low-

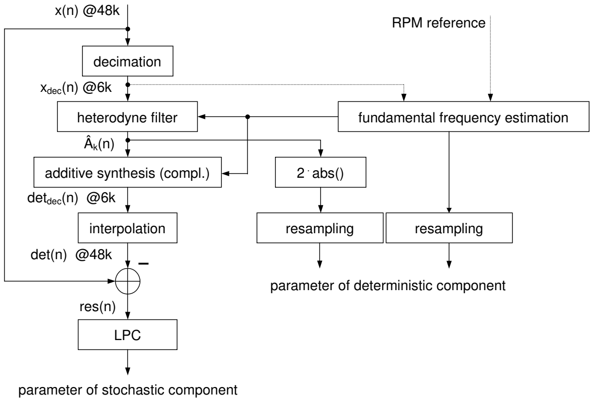

Figure 4: *The Analysis.*

pass filter $h_{LP}(n)$ should have zero phase. The choice of the filter bandwidth is very important. It should be as small as possible so that neighbouring sinusoids and noise don't influence the result. On the other hand, the bandwidth should be large enough to track fast changes in amplitude.

Vold in [4] proposed a different analysis method based on a state space formulation. Considering this Vold-Kalman tracking filter, we drew the conclusion that we can't take advantage of the decoupling of close or crossing partials, so it seems to be better to use alternative heterodyne filters which yield comparable results with less computing time (for example a zero-phase forward/reverse IIR filter).

Using the results of the heterodyne filters, we can obtain the isolated deterministic component using a (complex valued) additive synthesis:

$$det_{dec}(n) = 2\,\mathrm{real}\left(\sum_{k=1}^{K} \hat{A}_k(n) e^{j\varphi_k(n)}\right) \tag{6}$$

We interpolate this signal $det(n) = \mathrm{interpolate}(det_{dec}(n))$ to achieve the original sampling rate. Now the residual $res(n)$ can be calculated using subtraction in the time domain: $res(n) = x(n) - det(n)$. This subtraction is the reason for the zero-phase filter in (5) and the complex valued additive synthesis in (6), because it requires the phase coherency. We consider the residual as the stochastic component and use linear predictive coding (LPC) or FFT-based methods for modelling its spectral shape [3]. LPC provides filter coefficients for a filter of p^{th} order (reflection coefficients $\tilde{k}_i(m)$ for $i = 1, ..., p$ and gain $\tilde{G}(m)$).

3.3. The Resynthesis

The realtime resynthesis task is illustrated in figure 5. To convert the signal parameters, which are stored at a frame rate $\frac{1}{T_F}$ of about 20 Hz, to the original sampling rate we perform a linear interpolation.

$$v'(n) = v\Big(\Big\lfloor \frac{n}{H} \Big\rfloor\Big) + \left(v\Big(\Big\lfloor \frac{n}{H} \Big\rfloor + 1\Big) - v\Big(\Big\lfloor \frac{n}{H} \Big\rfloor\Big) \right) \frac{n - \lfloor \frac{n}{H} \rfloor H}{H} \tag{7}$$

v in equation (7) represents an arbitrary parameter of the signal model and H is the hopsize at original sampling rate ($T_F = H\,T$). This interpolation causes a latency of one frame interval T_F. We

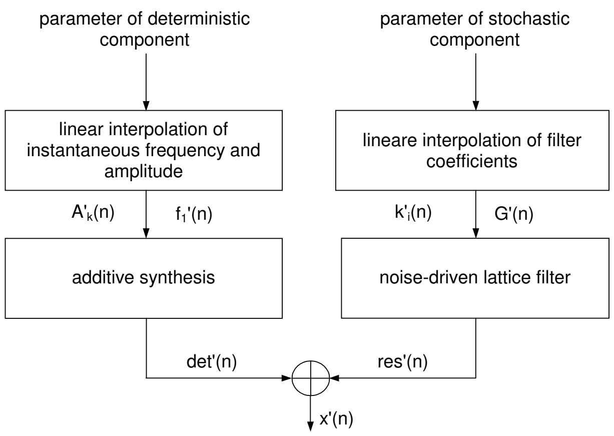

Figure 5: *The Resynthesis.*

calculate the deterministic component using a simple real valued additive synthesis:

$$det'(n) = \sum_{k=1}^{K} A'_k(n) \cos\left(2\pi \sum_{u=0}^{n-1} \big(k\, f'_1(u) + jit(u)\big)\, T \right) \tag{8}$$

We recommend to add a jitter $jit(n)$ to the instantaneous frequency which produces a slight inharmonicity to compensate the loss of phase information. This leads to more convincing results especially for low RPM.

For the resynthesis of the stochastic component, white noise is filtered either using an FFT-filter like in SMS or a time varying lattice filter.

4. CONCLUSIONS

We proposed an analysis/resynthesis system for the interior noise of vehicles which only needs a sparse set of parameters: the fundamental frequency trajectory, the real valued instantaneous amplitudes of the K sinusoidals and time varying filter coefficients. As a realistic example ($K = 30$ and $p = 30$) only 62 values per frame have to be stored representing 50 ms of the time signal.

Furthermore a very simple model for the load situation of a vehicle was introduced, which allows the interpolation between sounds at different accelerator pedal positions. Finally an algorithm for a realtime simulator of the interior noise of cars was presented.

Sound examples can be found at `http://iem.kug.ac.at/~feldbauer/`.

5. REFERENCES

[1] Serra, X., “A System for Sound Analysis/Transformation/Synthesis based on a Deterministic plus Stochastic Decomposition”, Dissertation, Dep. of Music, Stanford University, 1989.

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