Introduction of a novel analysis method to predict the performance of secondary systems

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ABSTRACT

Nowadays active noise control (ANC) systems are used in many different fields of applications. These applications include earth moving machines, cars, trains or airplanes, just to name a few. Thereby one is often confronted with the problem, that an already installed playback or sound system has to be used as a secondary system to fight the undesired noise.

Taking the car as an example, there it is the case that the sound system which incorporates the already installed speakers, has to be used to create zones-of-silence at one or more positions within the interior of the automobile.

Knowing about the theoretically achievable performance of this secondary system would be beneficial in assessing the operational spectral range of the ANC-system.

In this presentation, a method will be disclosed which allows the realistic making of predictions of the limitations concerning the secondary system, to which we are bound. Thereby, the only presumption is that a perfect primary system is expected, i.e. reference sensors delivering signals showing a perfect coherence to the signals picked up by the utilized error microphones.

The prediction is based on secondary path measurements done within the desired zones-of-silence, which ideally should be located close to the foreseen error microphone positions.

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1. INTRODUCTION

One vivid field of investigation and application of ANC systems is given in automotive environments.

There, in principle we distinguish between two main streams. The one, commonly known as an engine order cancellation (EOC) system, which denotes a narrowband feedforward ANC system, applicable in the interior, exhaust and/or heating, ventilation and air conditioning (HVAC) systems, is already used in some cars. The other, often referred to as a road noise cancellation (RNC) system, designates a broadband feedforward ANC system, preferably made to reduce noise in the interior of vehicles, is currently on its way to be integrated into automobiles, for the first time.

Since RNC systems are more complex and hence more complicated to be handled then EOC systems, especially in the former case, more enhanced analysis tools are needed to get a deeper insight into the system, in which noise shall be adaptively controlled.

A well known, important and often practiced method, used in broadband feedforward ANC systems, is the multiple coherence analysis (MCA) [1]. With the help of the MCA method, it is possible to make predictions about the quality of the used reference signals, with respect to the utilized error signal.

In an automobile environments we are faced with a special case, in which the positions of the error microphones are more or less known in advance, since we want to achieve a well perceivable damping performance, which means the error sensors have to be placed close to the listener’s head, which on their part, are defined by the known seat positions. In short, the error microphones have to be installed in the proximity of the listener’s head.

This means that the only remaining possibility to influence the resulting coherence between the reference and the error signals, which in turn provide insight to the maximal possible damping, is to find optimized positions for the reference sensors. But if this predicted damping is indeed to be achieved with the help of the MCA method, this also depends on the quality of the playback system.

The previously mentioned playback system consists not just of the secondary sources, installed in

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the interior of the car; this means its installed loudspeakers, but also of its positions with respect to the error sensors [2].

In this case, our possibilities to influence the playback systems are even more limited than before, since also the positions of the loudspeakers, installed in the car are defined a priori, this time by the car manufacturer. Therefore, the only chance we have is to choose the best fitting speakers, or to propose better suiting positions, for the purpose of increasing the performance of a broadband ANC system.

Therefore, another analysis method is needed, which is able to predict the performance of the playback, respectively the secondary system. The purpose of this paper is to disclose such an analysis method suited for this task.

2. METHOD

As was found by the author, the task of creating personal, respectively individual sound zones (ISZ) can be related to the RNC problem [3]. Thereby, also a similar signal processing framework, to be more precise, a multiple input multiple output (MIMO) system, utilizing, for example a multiple error filtered input least mean squared (MEFXLMS) algorithm, as introduced e.g. by [4], can be used to find the best fitting finite impulse response (FIR) filters, able to increase the acoustical contrast at certain zones, or, as in the RNC case, to enhance the zone(s) of silence within the room.

In contrast to the MCA method, in which it was intrinsically assumed that the secondary system is perfectly working, now, by utilizing one of the known ISZ methods (see e.g. [5]) to analyze the quality of the used secondary system, it is hypothesized to have perfect coherence between the signals of the reference and the error sensors.

Hence, by combining both analysis results we can find out to what extent the utilized secondary system is able to realize the maximum damping, as predicted by the MCA.

In the following, the basic principle, on which the RNC as well as the ISZ algorithm is based on, will be described.

2.1 Basic Principle

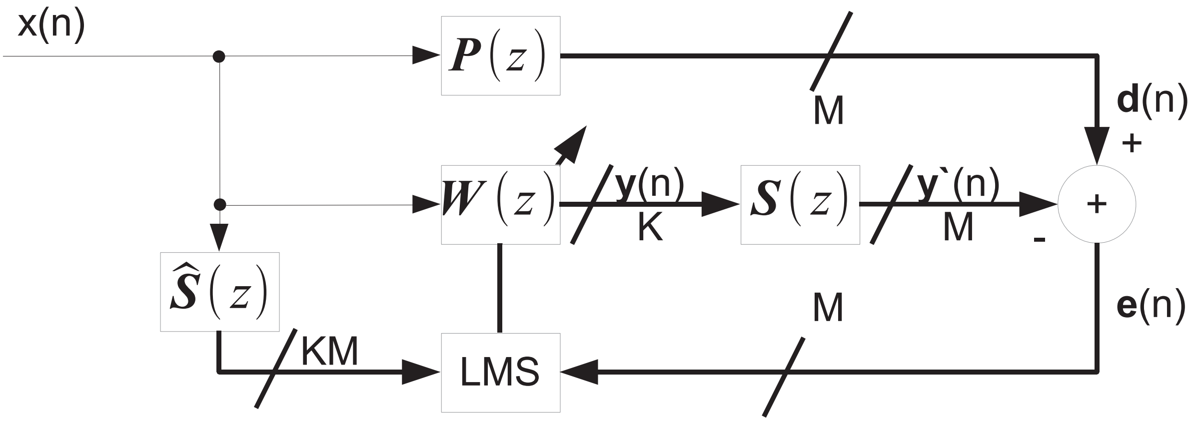

Figure 1 - Generalized $\mathbf{1\ x\ K\ x\ M}$ MIMO system, utilizing the MEFXLMS algorithm

The signal flow diagram (SFD) of Figure 1 shows the basic principle of the MEFXLMS algorithm, in which only one input signal x(n) is used as a reference. The acoustical reference signal x(n) is filtered by the (acoustical) primary path filter matrix P(z). Thereby, the M desired signals d(z) are picked up, together with the M signals y′(n), by the M error microphones and thus result in the M error signals e(z). The K loudspeaker driving signals y(z) arise by filtering the (electrical) reference signal x(n) with the K ANC filter W(z), which will be adapted to minimize the multiple error signals e(z). The signals y(z) are acoustically radiated by the K loudspeakers and filtered by the (acoustical) secondary path filter matrix S(z), before the resulting signals y′(n) overlay with the desired signals d(z), in the acoustical domain and result in the electrical error signals e(z) as received by the error microphones.

In addition, the input signal x(n) will be filtered by a K x M matrix, approximating the complete set of secondary path transfer functions between the K loudspeakers and the M error microphones, resulting in the K x M filtered input, respectively reference signals x′(n) (not shown in Figure 1). The K x M filtered input signals x′(n) and the M error microphone signals e(z) constitute the inputs to the least mean squared (LMS) block, in which the update of the K ANC filter $W(z)$ happens.

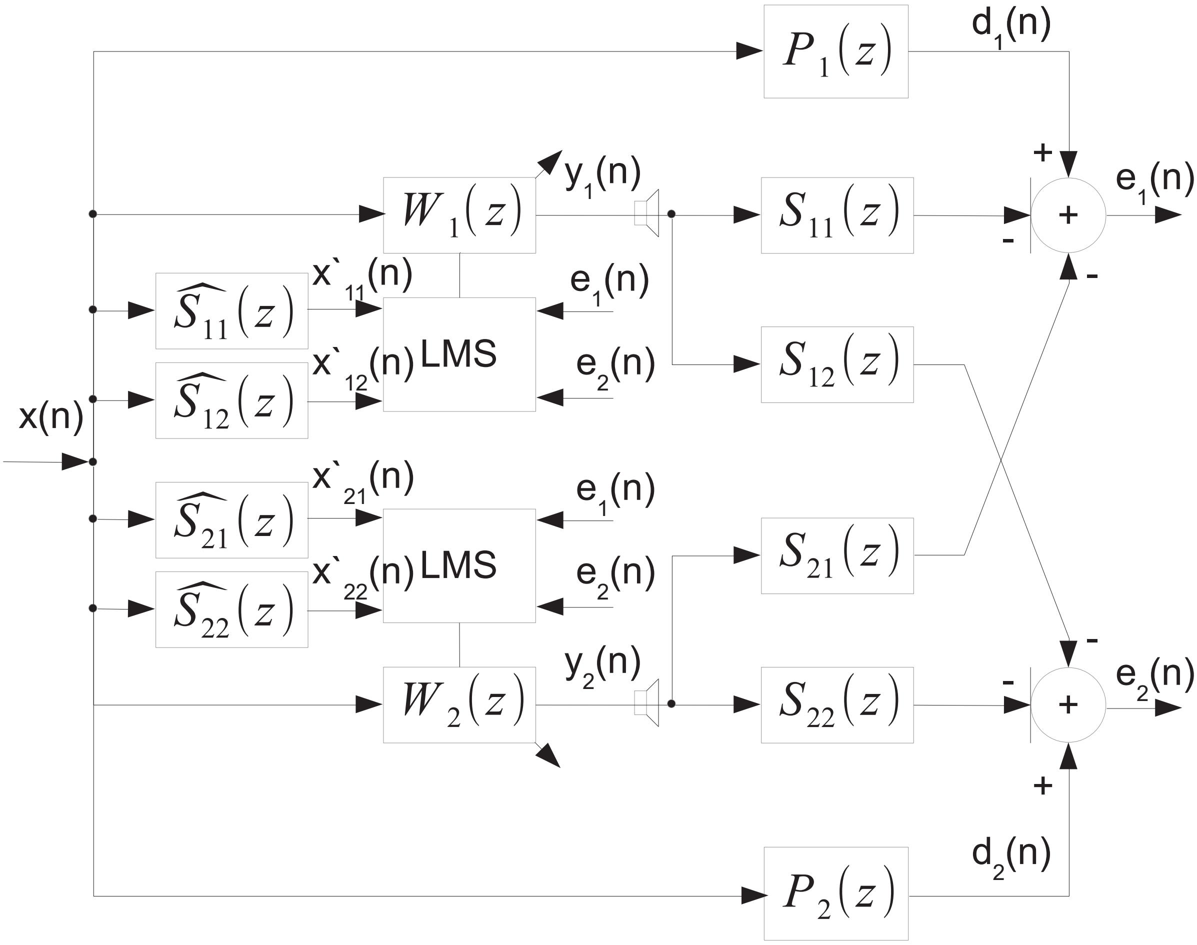

Figure 2 – $\mathbf{1\,x\,2\,x\,2}$ MIMO system, utilizing the MEFXLMS algorithm

Figure 2 depicts a $1\,x\,2\,x\,2$ MIMO system in more detail. Here, the position of the secondary sources, i.e. of the $K = 2$ loudspeakers, can be seen within the MEFXLMS algorithm. In addition also all individual secondary paths are displayed in the acoustical domain ($S_{k,m}(z)$, with $k, m \in [1, \dots, 2]$) as well as in the electrical domain ($\hat{S}_{k,m}(z)$, with $k, m \in [1, \dots, 2]$). The latter are used as filters approximating its acoustical counterparts, with which the reference signal $x(n)$ has to be weighted, before the resulting $2 \, x \, 2$ filtered input signals ($x'_{k,m}(n)$, with $k, m \in [1, \dots, 2]$) result. Together with the $M = 2$ error signals ($e_m(n)$, with $m \in [1, \dots, 2]$) the filtered reference signals $x'_{k,m}(n)$ are inserted into the LMS algorithm in order to update the $k = 2$ ANC filter ($W_k(z)$, with $k \in [1, \dots, 2]$). All contributing primary path filter ($P_m(z)$, with $m \in [1, \dots, 2]$) are evident in Figure 2, too. Since only one reference signal is used, the size of the primary path filter matrix $P(z)$ is reduced to $1 \, x \, M$.

In the spectral domain, mathematically the MEFXLMS algorithm can be described as follows:

Input-, respectively reference signal $X(e^{j\Omega}, n)$:

$$X(e^{j\Omega}, n) = FFT\{x(n)\} \quad (1)$$
$$x(n) = [x(nL), \ldots, x(nL+N)]^T$$

Where:

$x(n)$ = Vector of the reference signal of length N,
L = Feed, respectively frame shift,
N = FFT length.

Filtered input-, respectively reference signal $X'_{k,m}(e^{j\Omega}, n)$:

$$X'_{k,m}(e^{j\Omega}, n) = X_{k,m}(e^{j\Omega}, n)\,\hat{S}_{k,m}(e^{j\Omega}, n) \quad (2)$$

Where:

k = $1, \ldots, K$,
K = Number of loudspeakers,
m = $1, \ldots, M$,
M = Number of error microphones,
$\hat{S}_{k,m}(e^{j\Omega}, n)$ = Approximation of the secondary path between the k^{th} loudspeaker and the m^{th} error microphone.

Loudspeaker driving signals $y_k(n)$:

$$y_{k,Comp}(n) = IFFT\{X(e^{j\Omega}, n)\,\widetilde{W}_k(e^{j\Omega}, n)\}, \forall\, k = [1, \ldots, K] \quad (3)$$
$$y_k(n) = [y_{k,Comp}(N-L+1), \ldots, y_{k,Comp}(N)]$$

Where:

$\widetilde{W}_k(e^{j\Omega}, n)$ = Constraint adaptive filter, applied to the k^{th} loudspeaker.

Secondary signals, picked up by the M error microphones $y'(n)$:

$$y'_m(n) = \sum_{k=1}^{K} \left(y_k(n) * s_{k,m}(n)\right), \forall\, m = [1, \ldots, M] \quad (4)$$

Where:

$*$ = Convolution operator,
$s_{k,m}(n)$ = Impulse response of the secondary path between the k^{th} loudspeaker and the m^{th} error microphone.

Error signals $e(n)$, respectively $E(e^{j\Omega}, n)$:

$$e_m(n) = \sum_{m=1}^{M} \left(x(n) * p_m(n) - y'_m(n)\right), \forall\, m = [1, \ldots, M] \quad (5)$$

Where:

$p_m(n)$ = Impulse response of the primary path between the (first) input signal $x(n)$ and the m^{th} error microphone.

$$E_m(e^{j\Omega}, n) = FFT\left\{\begin{bmatrix} 0 \\ e_m(n) \end{bmatrix}\right\}, \forall\, m = [1, \ldots, M] \quad (6)$$

Where:

0 = Column vector, filled with zeros of length N-L,
$e_m(n)$ = Vector holding samples of the m^{th} error microphone of length L.

Adaptation, respectively update of the K adaptive filters $W_k(e^{j\Omega}, n)$:

$$W_k(e^{j\Omega}, n+1) = W_k(e^{j\Omega}, n) + \mu \sum_{m=1}^{M} \left(X'_{k,m}(e^{j\Omega}, n)\, E_m(e^{j\Omega}, n)\right), \forall\, k = [1, \ldots, K] \quad (7)$$

μ = Adaptation step size.

Constraint:

$$
\begin{aligned}
\overline{\mathrm{w}}_{\mathrm{k,Comp}}(\mathrm{n}+1) &= \mathrm{IFFT}\{\mathrm{W_k}(\mathrm{e}^{\mathrm{j}\Omega},\mathrm{n}+1)\}, \forall\, \mathrm{k}=[1,\dots,\mathrm{K}] \\
\overline{\mathrm{w}}_{\mathrm{k}}(\mathrm{n}+1) &= [\overline{\mathrm{w}}_{\mathrm{k,Comp}}(0),\dots,\overline{\mathrm{w}}_{\mathrm{k,Comp}}(\mathrm{N}-1)] \\
\widetilde{\mathrm{W}}_{\mathrm{k}}(\mathrm{e}^{\mathrm{j}\Omega},\mathrm{n}+1) &= \mathrm{FFT}\left\{\begin{bmatrix}\overline{\mathrm{w}}_{\mathrm{k}}(\mathrm{n}+1)\\ 0\end{bmatrix}\right\}
\end{aligned}
\tag{8}
$$

Where:

$\overline{\mathrm{w}}_{\mathrm{k}}(\mathrm{n}+1)$ = Filter coefficients of the k^{th} adaptive filter of length $\mathrm{N}/2$.
0 = Column vector, filled with zeros of length $\mathrm{N}/2$,

If we want to create a bright zone at the position where error microphone 1 is located in the room, and a dark zone at the position of error microphone 2, the primary path $\mathrm{P_1(z)}$ has to be replaced by a corresponding modeling delay, as known e.g. from [2], whereas the primary path $\mathrm{P_2(z)}$ has to be set to zero, as illustrated in Figure 3.

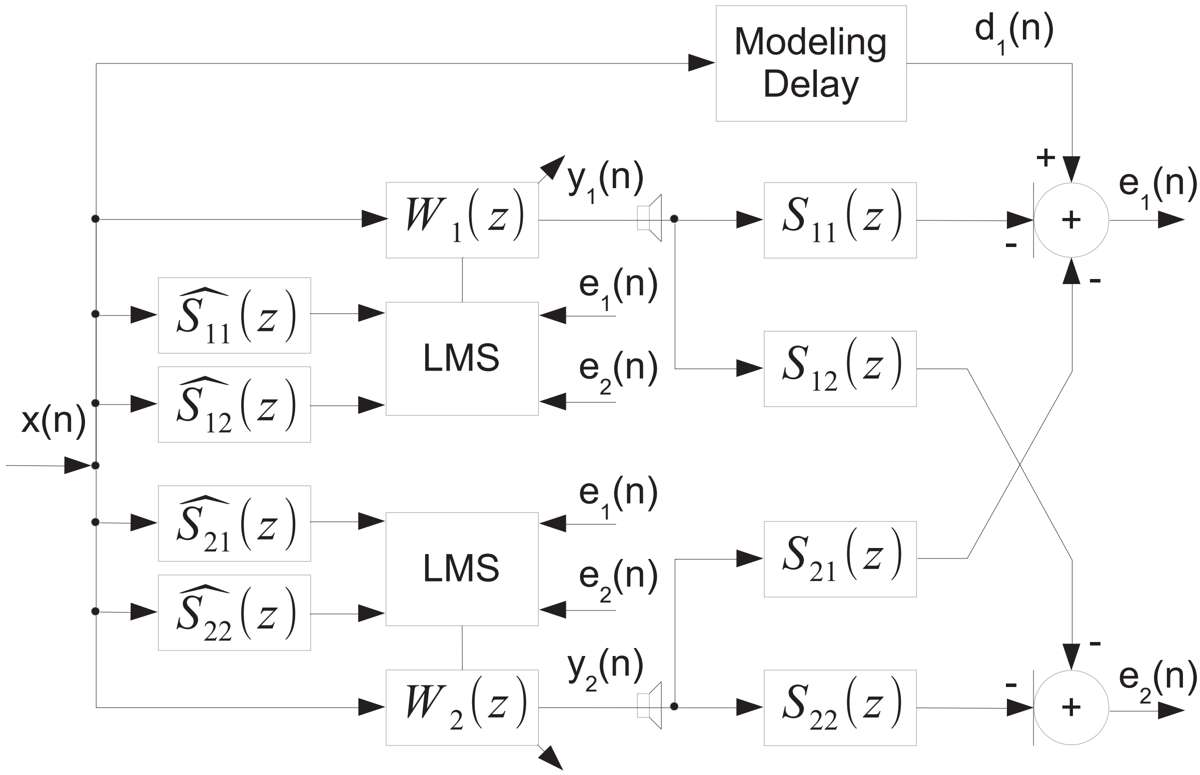

Figure 3 – Modified MEFXLMS algorithm for the creation of individual sound zones (bright zone at error microphone 1 ($\mathbf{e_1(z)}$) and a dark zone at error microphone 2 ($\mathbf{e_2(z)}$))

Whilst the modified MEFXLMS algorithm of Figure 3 is active, the error signal $\mathrm{e_1(z)}$ converges toward a time delayed version of the reference signal $\mathrm{x(n-N/2)}$, assuming that an optimal modeling delay of length $\mathrm{N/2}$ was used. At the same time the error signal $\mathrm{e_2(z)}$ converges towards zero.

As a result, at the position of error microphone 1, a signal, ideally corresponding to the reference signal $\mathrm{x(n)}$ can be observed, i.e. a bright zone will be established where error microphone 1 is located in the room, while, at the same time a dark zone will be created at the position of error microphone 2.

If we set the primary path $\mathrm{P_1(z)}$ of Figure 3 to zero, the system could also be used to create two dark zones, which would, at least to some extent resemble the RNC case. The algorithm then converges in such a way that now at both locations of the error microphones dark zones, respectively zones of silence will be created, at which the known, single primary source would be suppressed as much as possible, utilizing the given secondary system.

In practice, there is not just a single (known) primary source which a RNC system has to deal with, but many, which signals are tried to be measured, respectively estimated with the help of reference sensors, such as microphones or accelerometers (ACC's), just to name a few.

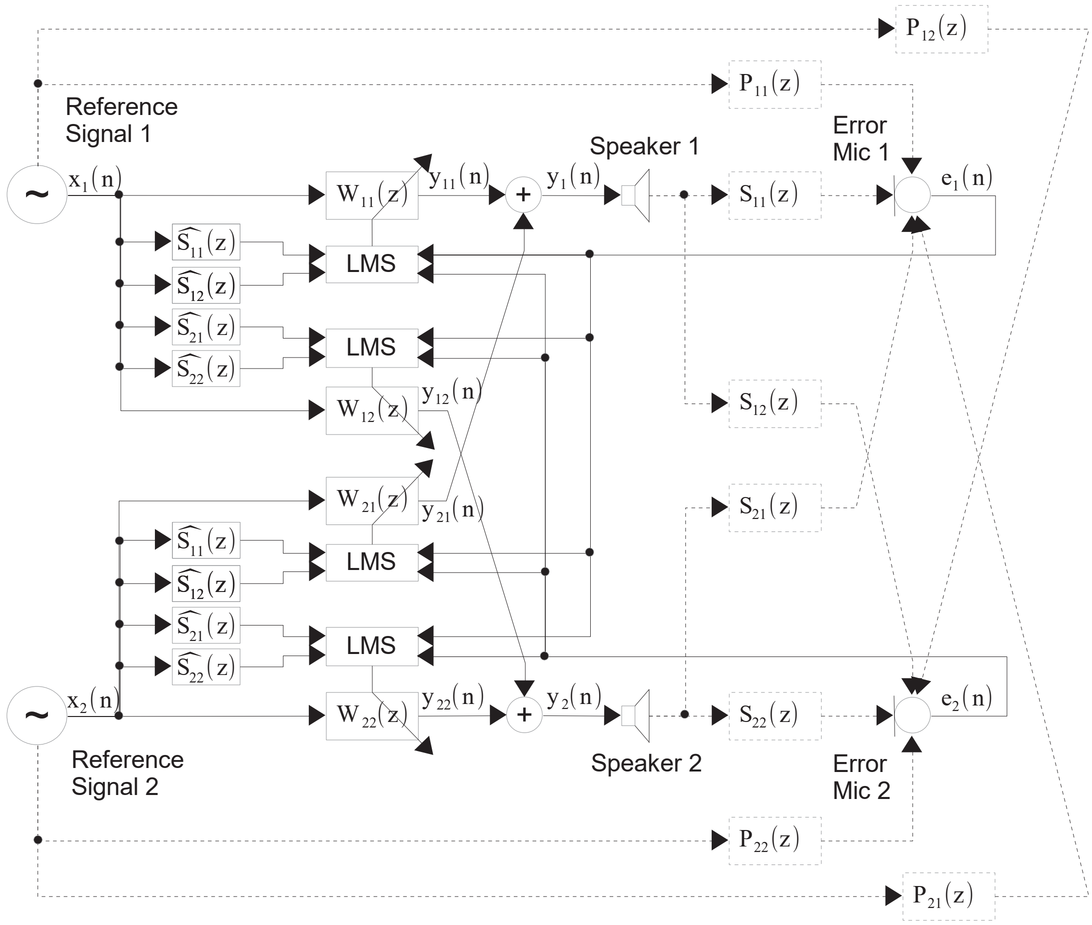

Figure 4 – $\mathbf{2\ x\ 2\ x\ 2}$ MIMO system, utilizing the MEFXLMS algorithm, shown in more detail

Figure 4 illustrates a typical RNC situation, exemplarily shown on a $\mathrm{I\,x\,K\,x\,M}$ system, with $\mathrm{I} = \mathrm{K} = \mathrm{M} = 2$. For this most simple multichannel example, not just one, but two reference signals $(\mathrm{I} = 2)$ are used as inputs, originating e.g. from ACC sensors, mounted on well suited positions at the body of the car. Like before, $\mathrm{K} = 2$ loudspeakers and $\mathrm{M} = 2$ error microphones were used. Now, a total of $\mathrm{I\,xM}$ primary paths $\mathrm{P_{i,m}(z)}$, with $\mathrm{i} = [1, \dots, \mathrm{I}]$ and $\mathrm{m} = [1, \dots, \mathrm{M}]$ and $\mathrm{I\,x\,K}$ control filters $\mathrm{W_{i,k}(z)}$, with $\mathrm{i} = [1, \dots, \mathrm{I}]$ and $\mathrm{k} = [1, \dots, \mathrm{K}]$ exist, where the latter have to be adapted.

Furthermore, in Figure 4 acoustical paths are accentuated by dashed lines, whereas electrical signals are drawn with solid lines, for better understanding.

Comparing Figure 3 with Figure 4, it is easy to recognize the structural similarity of the signal processing framework that can be used for the creation of ISZ filter, as well as for a RNC system.

3. VERIVICATION

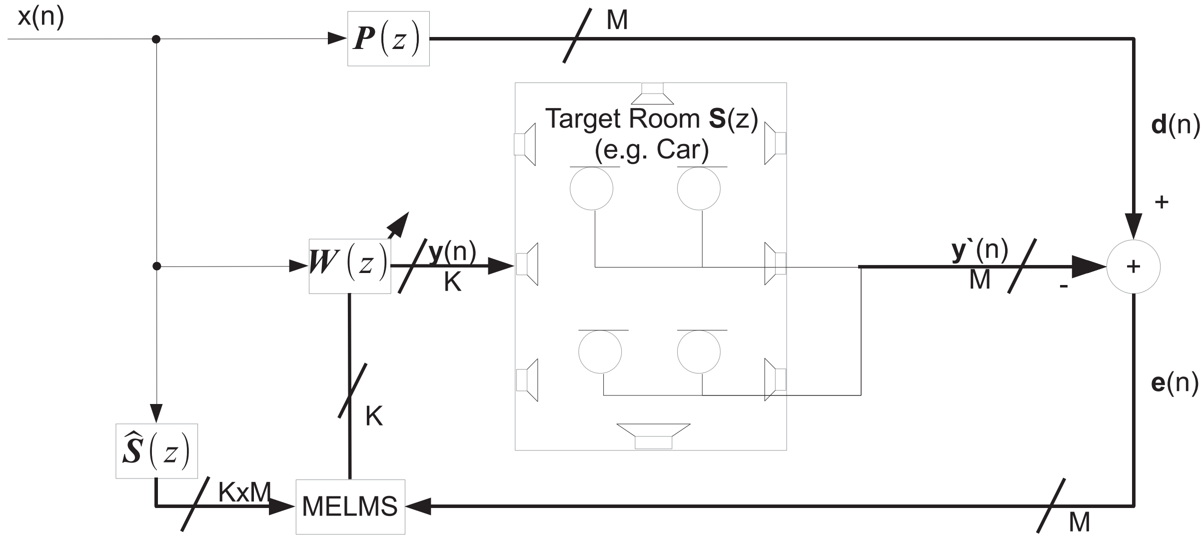

Figure 5 – $\mathbf{1\ x\ 2\ x\ 2}$ MIMO system, utilizing the MEFXLMS algorithm, shown in more detail

In this chapter, the idea and theory of the novel method for analyzing secondary systems, which were introduced in the previous chapters, will be verified. For this reason the secondary path of an exemplary car were measured and inserted in an MEFXLMS algorithm, as shown in Figure 3.

Figure 5 depicts a representative situation, which can be found in many cars. Thereby, the positions of the error microphones, typically mounted in the headliner, in proximity to the head locations of the passengers in the interior of the car, can be seen, as well as characteristically loudspeaker positions, used in automotive sound systems.

One goal of the novel analysis method is to find out which of the already installed speakers can be used beneficially. Of course it would always be best to use as many speakers as possible, but in reality there are certain limitations, such as limited processing power of the utilized processor and/or a restriction in memory. Thus, in terms of cost and energy consumption etc, it is always preferable to keep the whole system as small as possible – ideally without sacrificing damping performance.

Before we start with the analysis of the secondary system, utilizing the previously described ISZ framework, we will take a look at a typical result of an MCA.

Thereby in total four two-axis ACC sensors, arranged along the y- and z-axis, were mounted at the chassis of the car. More precisely, two of the four two-axis ACC sensors were mounted on the front subframe at its left and right positions, whereas the remaining two ACC sensors were mounted at both rear hubs – again one mounted on the left and the other on the right hand side.

The MCA was then calculated with respect to all four error microphones, installed close to all four head, respectively seat positions.

If we draw a line at a coherence vale of $\gamma_{k,m} = 0.5$, which corresponds, regarding the below formula (see e.g. [6], page 930), to a maximally achievable damping of $NR_{dB} \sim 3\ [dB]$, we find a maximal controllable spectral range of up to $f_{max} \sim 350\ [Hz]$ for this car and the utilized ACC sensors, mounted at the specified positions.

$$\mathrm{NR_{dB}} \quad = \quad 10 * \log\Big(1/\big(1 - \gamma_{k,m}\big)\Big) \tag{9}$$

Where:

NR_{dB} = Noise reduction in $[dB]$,

$\gamma_{k,m}$ = Mean squared coherence (MSC) between the kth and the mth signal.

As a result, it can be stated that the playback system should be able to cover at least this spectral range in order to achieve there the possible damping, as predicted by the MCA (see Figure 7), utilizing the specified ACC sensors as well as the same error microphones.

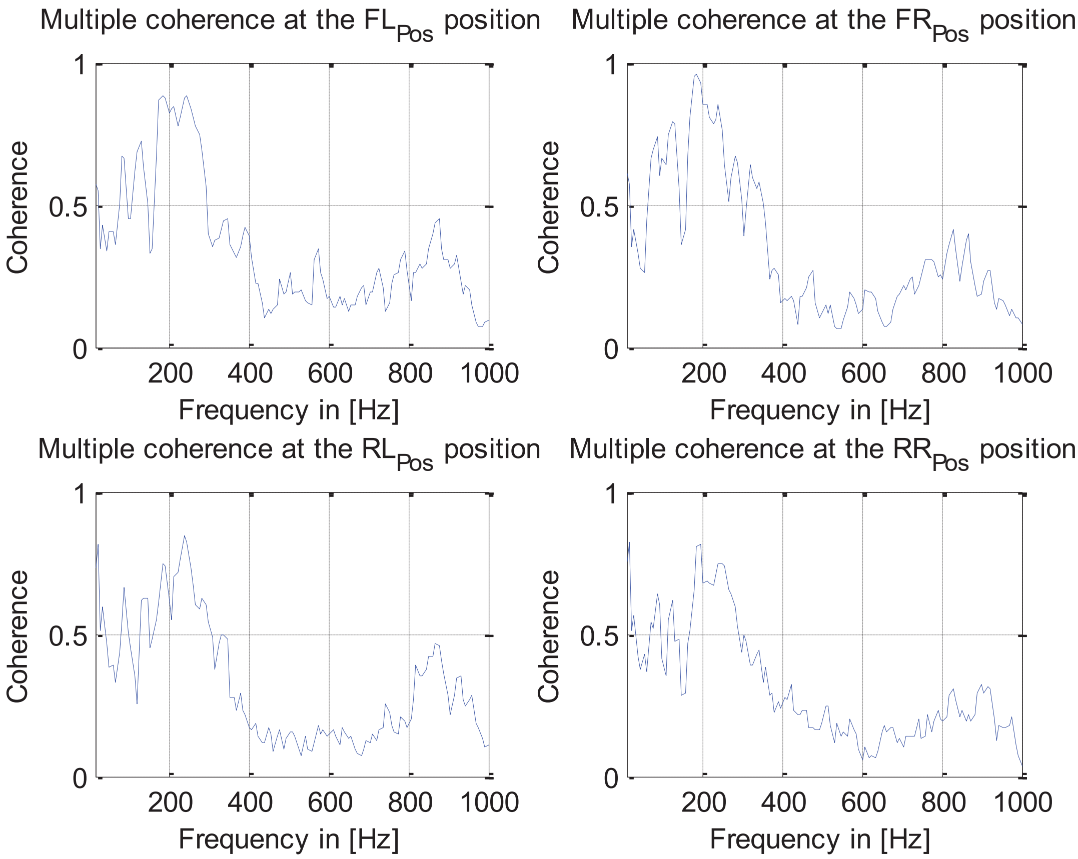

Figure 6 –MCA at all four seat positions, where the corresponding error microphones had been installed, utilizing the following ACC sensor signals: $\boldsymbol{FL_{SubY}}$, $\boldsymbol{FL_{SubZ}}$, $\boldsymbol{FR_{SubY}}$, $\boldsymbol{FR_{SubZ}}$, $\boldsymbol{RL_{HubY}}$, $\boldsymbol{RL_{HubZ}}$, $\boldsymbol{RR_{HubY}}$, $\boldsymbol{RR_{HubZ}}$ for the calculation of the multiple coherence.

Due to the result of the MCA, that only at a limited spectral range of up to f_{max}~$350\ [Hz]$ can noteworthy damping be achieved with the utilized ACC and error sensors, which can be seen from Figure 7, for the following analysis of the secondary system, only a reduced loudspeaker set, including both front and rear door woofers as well as the subwoofer, installed in the trunk was used, since only these five speakers met the prerequisites to deliver sufficient sound pressure level (SPL) within this limited spectral range.

In addition, for the analysis of the secondary system, the ISZ system was adjusted in such a way that a bright zone was created at the error microphone, corresponding to the driver's position (FL_{Pos}), whereas at all other error microphone locations, corresponding to the remaining three seat positions (FR_{Pos}, RL_{Pos}, RR_{Pos}) dark zones were established. Thereby the target magnitude frequency response at the driver's position was deliberately set to a flat line to enable an easier reading of the acoustical contrast, given by the differences between the magnitude frequency responses of the dark zones and the one measurable at the bright zone.

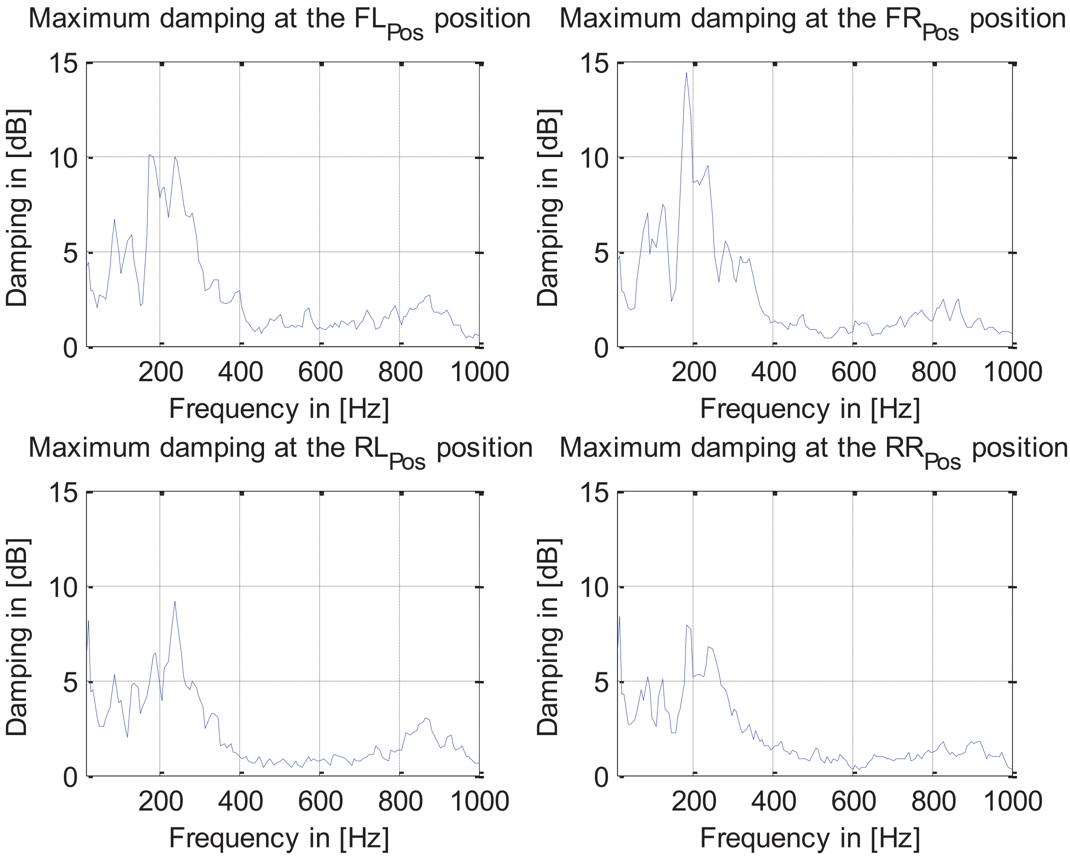

Figure 7 – Damping performance at all four seat positions, as predicted by the MCA.

Taking a look at Figure 8 we can see, by comparing these results with the ones shown in Figure 7, that the specified secondary system, respectively the chosen loudspeakers of the car, are generally able to generate enough damping within a spectral range of $f_{Rear} \sim [180, \dots, 350]\ [Hz]$ at the rear seats and in a frequency range of $f_{Front} \sim [100, \dots, 350]\ [Hz]$ for both front seats.

Generally it can be stated that coherence levels of $\gamma_{k,m} < 0.7$ are usually not sufficient to allow the MEFXLMS algorithm to converge.

Hence in this example we don't have to expect severe problems, since the utilized reference and playback systems match together relatively well, which is not always the case.

Figure 8 also reveals the limitations of the specified playback system, which is reached as soon as no more damping (difference between bright and dark zones) can be generated. In this example this upper frequency limit can be shown to be at an upper frequency of $f \sim 1[kHz]$.

The somewhat less damping of the front seat passenger and both rear seat positions can be explained with the choice to generate the bright zone at the front left seat. It is a well known fact in ISZ systems that neighboring seats do usually not have as much acoustical contrast as positions at different rows, but this does not mean that this is also the case in RNC applications. Thus, we should concentrate on the acoustical contrast achievable at different rows, since these values are better suited for the analysis of a playback system, applicable for RNC systems.

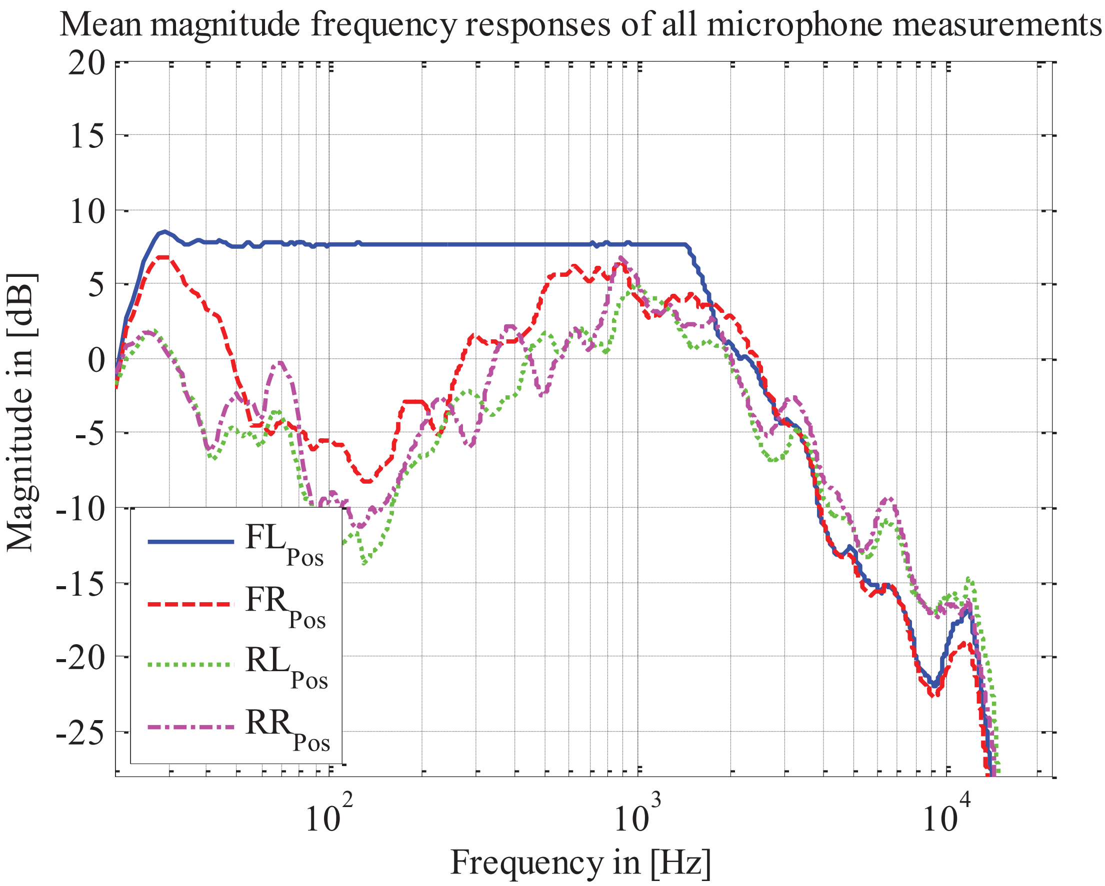

Figure 8 – Maximum damping which can be achieved by the utilized secondary system.

(The achievable damping can be identified by taking the difference of the shown magnitude frequency responses, that can be measured at different positions in the car. It is clearly perceivable that a bright zone is created at the front left position (blue line) while at the remaining seat positions dark zones are generated.)

4. CONCLUSION

This paper showed to what extent the concept to create individual sound zones can be utilized as novel analysis tool for the judgment of the playback systems which are predetermined most of the time, as is the case in cars for example. This information about the ability of the secondary system is necessary to get a more complete picture of the involved components of a broadband ANC system.

In the past, only the MCA was used to obtain information about the possible damping performance of the reference system, consisting of the reference sensors as well as of the error sensors. Thereby, the multiple source analysis utilizing the MCA method is a suitable and valid tool for source identification, i.e. to find well performing inputs for the ANC system, thereby implicitly assuming a perfect functioning secondary system. No doubt, this is of major importance, since only those noise components that can be measured can be cancelled.

Otherwise, even if there are perfectly decorrelated input signals (which is assumed in the here disclosed, novel analysis method for secondary systems) this does not automatically mean that these perfect conditions of the reference system are reflected 1:1 in the eventually perceivable damping performance. The de facto perceivable damping performance also depends on the damping performance of the secondary system. Thereby, the damping performance of the secondary system has to at least match, or preferably outperform the predicted damping performance known from the MCA. If this is not the case, recommendations based on these analysis results can be provided to the car manufacturer, in order to improve as an example the location of the existing speakers, replace them with ones better suited, and/or to propose additional speakers at appropriate locations in the interior of the car.

As a result, if the analysis shows that the damping performance of the secondary system does not at least match the predicted damping performance of the reference system, no perceivable increase of the damping performance can be achieved.

Hence, it can be stated that the analysis of the secondary system is almost as important as the analysis of the reference system to eventually make valid predictions of the finally perceivable damping at difference locations in the room.

Thus, it is strongly hypothesized that the analysis of the secondary system, e.g. by utilizing the ISZ method, as proposed in this paper, will become a standard tool in the analysis of ANC systems, in the future.

Simulations based on baseline measurements include all limitations and as such deliver a viable estimation of the damping that can be expected, at least at the error microphone positions where the measurements were conducted. Certainly, such a simulation also provides useful information of the whole ANC system, but it fails to deliver important specifics of certain aspects of the complete system, essential e.g. if the real, respectively the simulated system fails or delivers a high damping performance which is not as expected. In this case, additional analysis methods, such as the MCA or the proposed ISZ based concept, are needed.

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