CONTROL ALGORITHMS FOR LOCATING ZONES OF QUIET IN THE ACTIVE HEADREST SYSTEM*

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A system known as the active headrest, generating the so-called zones of quiet at the ears of a person sitting in a chair is considered. Two adaptive control algorithms using the idea of virtual microphones are designed. One of them, guaranteeing very high attenuation, is advised if the noise is stationary. Another control system proposed has much more general character and can be successfully applied for both stationary and non-stationary noises. It, in turn, assumes that the primary source is far from the headrest compared to the acoustic wavelength. Obtained results are presented as distribution of attenuation areas.

Keywords: active noise control, active headrests, adaptive feedback control.

1. INTRODUCTION

In addition to loss of concentration and annoyance, many people suffer from severe hearing damage due to high-level ambient noise in their working environments. Prolonged exposure to loud sounds causes damage to the hair cells with the result that hearing ability becomes progressively impaired. Besides, as recent investigations show, it has also negative influence on other basic human systems.

Despite the great progress in technology, it is still very difficult or impossible to produce "silent" machines. Also, in many practical situations, passive barriers are not adequate. They are very costly or difficult to design when the worker must be in a close

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contact with a noisy machine. In such cases active noise control (ANC) devices are advised to be used as an interim solution.

In ANC, in general, additional secondary sound sources are used to cancel noise from original primary sources. The physical justification is given by Young's interference principle. This theory, although formally very simple, is difficult to be directly applied in practice. There are many problems related to physical aspects of the cancellation phenomenon as well as related to control.

It could be proved that in a diffuse field every acoustic mode generated should be controlled by one secondary source. The number of modes is proportional to the room volume and third power of noise frequency [4]. Hence, global noise control is practically unfeasible. The solution is thus local control at some particular areas and creating the so-called zones of quiet. This idea is successfully used in active headsets that generate "silence" directly inside the ears. However, active headsets are not comfortable devices and they are widely accepted by practitioners only if people are not assigned to stationary work places but need to move. Nevertheless, there are many professions that require staying at one place most of the time, e.g., in control rooms, at assembly lines, in caterpillar cabins, etc. Therefore, another solution should be offered.

This article concentrates on creating zones of quiet around ears of a person sitting in a chair. The headrest of a chair is equipped with loudspeakers generating artificial sounds as well as microphones sensing interference effects. Such a device is known in the literature as the active headrest system [8-10].

The original idea of actively controlling sound in a headrest has first been presented by Olson and May in1953 who applied a feedback control system . However, this idea needed to wait until the 1990s due to problems with phase lag in the audio amplifier to ensure stability of the loop [1]. Advances in microelectronics, high–speed signal processors, and filtering techniques during the 1980's precipitated a flurry of activity in digital control systems. The idea mentioned has been undertaken at the end of that decade by Rafaely, Elliott and Garcia-Bonito who thoroughly analysed both acoustical and control limitations, and gave recipes for fixed controllers design using H_2 / H_∞ approach [8-10]. They also analysed the zones of quiet generated and stressed that for significantly low frequencies the zones are large enough to reach human's ears but for higher frequencies they put forward the theory of virtual microphones allowing to shift the zones.

The main contributions of this article are designing and experimentally verifying two efficient adaptive control systems (adopting the idea of virtual microphones) that generate areas of highest attenuation at the human's ears. One of them takes advantage of potential noise stationarity to reduce it almost to the measurement noise level. The other is more general but assumes that the primary source is located far from the headrest. All algorithms considered are of feedback structure because in practice the primary source is usually distributed and therefore it is impossible to measure appropriate reference signal. Beside, this article offers a new standard for presenting active noise control results as distribution of zones of quiet providing a good insight into the cancellation process and allowing to evaluate practical meaning of control algorithms applied.

2 THE HEADREST SYSTEM

Laboratory rig used for the research reported is presented in Fig. 1. It consists of the headrest, i.e., a frame supporting the head with two loudspeakers (secondary sources), G1 and G2, accompanied by two electret error microphones, e1 and e2. Obviously, a loudspeaker (primary source, G3), generating the noise, DS1102 board with TMS320C31 signal processor and converters, a computer, power and voltage amplifiers, and a set of analogue low-pass filters are also necessary.

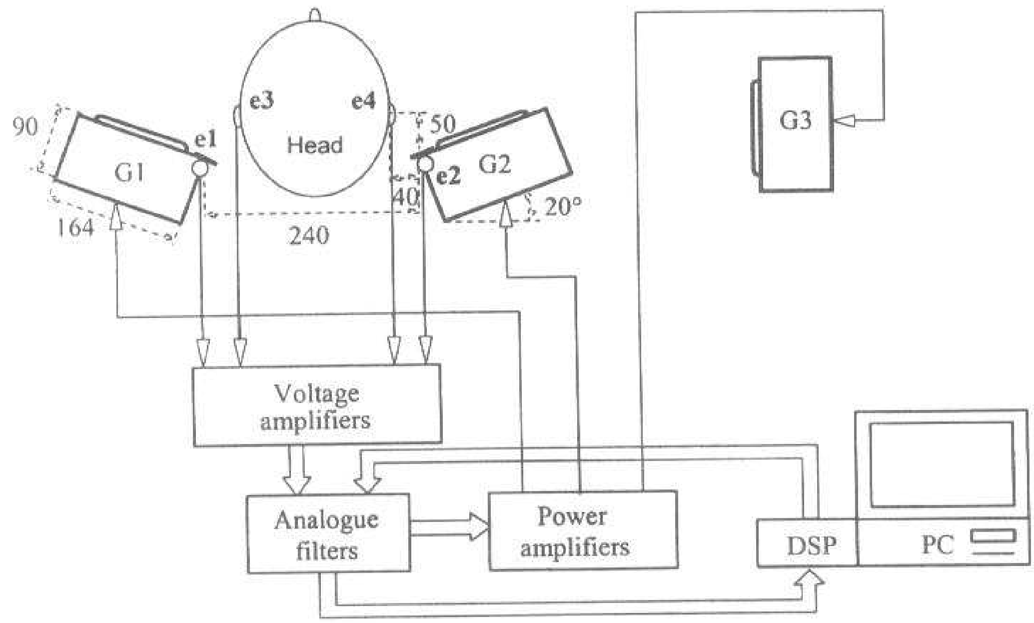

Fig. 1 The active headrest system with arrangement of loudspeakers and microphones.

Microphones e3 and e4, placed in locations where the attenuation is desired, i.e., at the human's ears, are used for performance monitoring and they are not employed by control algorithms during their operating stages.

A compact block diagram of the plant is presented in Fig. 2, where S_{11}, S_{22} are transfer functions of the main paths (from loudspeakers to error microphones in the same channels), and S_{12}, S_{21} are transfer functions of the cross paths (from loudspeakers to error microphones in neighbouring channels). Signals $d_1(i)$ and $d_2(i)$ are the discrete output disturbance signals, i.e., discretised primary noises. Using control system terminology the primary noise constitutes a disturbance to be attenuated.

For designing and parameterising control structures considered in this article it is important to know frequency responses and parametric FIR (finite impulse response) models of all paths from Fig. 2. However, due to complicated coupled acoustic and electric phenomena it is impossible to build sufficiently precise phenomenological models. Therefore, identification techniques were employed. The aspects of experimentally identifying electro-acoustic plants are discussed in details in [6]. The sampling frequency used to excite the plant inputs and measure its outputs was 2 kHz. Cut-off frequencies of the analogue 4th order Butterworth filters (anti-aliasing and reconstruction) were set to 650 Hz, providing sufficient signal reduction at the Nyquist frequency (half the sampling frequency) and not introducing excessive phase lags resulting in large discrete time delays. Magnitudes of frequency responses of exemplary main and cross paths are presented in Fig. 3.

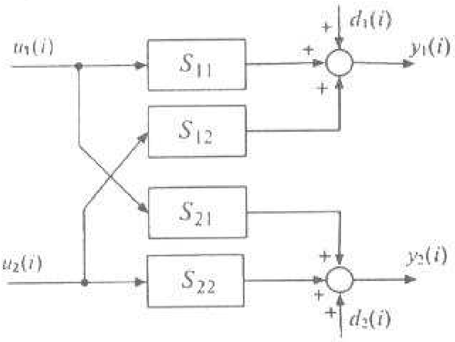

Fig. 2 The plant.

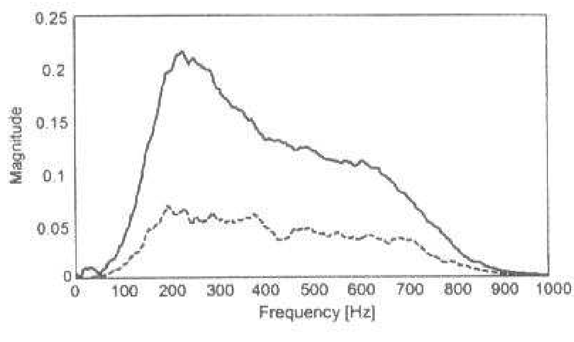

Fig. 3 Magnitudes of frequency responses of exemplary paths: S_{11} (solid) and S_{12} (dashed).

Comparing the responses of a main and cross paths, it is seen that contribution of cross paths to the system is significant and they should be taken into account in the design process. From other identification experiments also follows that the main paths reveal delays of 4 samples but the cross paths – of 6 samples [11].

3 CLASSICAL CONTROL STRUCTURE

Under assumption, all the algorithms can only be based on microphones e1 and e2 mounted in the headrest (see Fig. 1). Nevertheless, most well developed adaptive algorithms for active noise control giving best results require a reference signal [1], [2], [4]. For the sake of them, Internal Model Control (IMC) structure will be used, in which this signal can be estimated (see Fig. 4) [3].

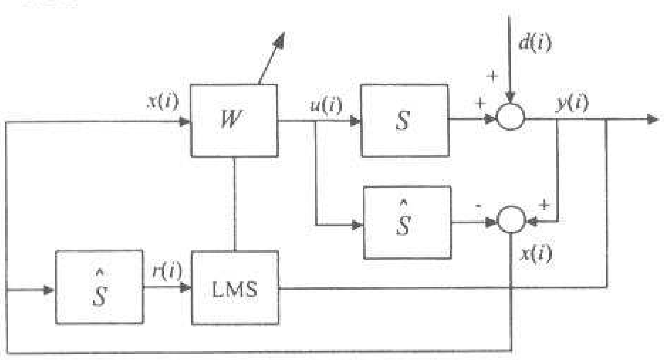

Fig. 4 IMC system with FXLMS algorithm.

In this structure the reference signal being the input to the control filter, W, driving the plant is calculated as:

$$x(i) = y(i) - \hat{\mathbf{s}} * \mathbf{u}(i), \tag{1}$$

where $\hat{\mathbf{s}}^T = [\hat{s}_0, \hat{s}_1, ..., \hat{s}_{L-1}]$ is the vector of parameters of $\hat{S}$ (FIR model of S – one of the paths from Fig. 2), generally time-varying if the plant response is subject to change, but * denotes linear convolution operator [2]. Theoretically, if the plant model is perfect, $\hat{S} = S$, the reference signal, $x(i)$, is a good estimate of the disturbance, $d(i)$, to be attenuated. Parameters, $\mathbf{w}^T = [w_0, w_1, ..., w_{L-1}]$ of the control filter, W, are estimated using well-known Filtered-x LMS (FXLMS) algorithm – see Fig. 4, [1], [2], [4]:

$$\mathbf{w}(i+1) = \mathbf{w}(i) - \mu(i)\left[\hat{\mathbf{s}} * \mathbf{x}(i)\right] y(i). \tag{2}$$

In (2) parameter $\mu(i)$ is the time-varying step size depending on modification of the basic LMS algorithm. In this research correlation LMS has been used, keeping tracking capabilities, reducing mean-square error (MSE) of the output in the steady-state and making the algorithm very stable [2].

Obviously, to generate zones of quiet at both ears, two channels should be put into operation (see Fig. 1). First experiments performed confirmed conclusions drawn from analysis of identification results in terms of mutual dependence of the channels due to the significant acoustic coupling (see Fig. 3). This interaction was particularly evident in adaptive systems, where contributing signals of the two channels "disturb" each other (for fixed controllers such unwanted effects can be marginal [8], [9]). Therefore, the adaptive IMC system in fully multidimensional version, two inputs – two outputs, has been employed [2].

The results for tones measured the by the error microphone (e1, e2) are presented in Fig. 5 but measured by the observer microphone (e3, e4 – at the ear) – in Fig. 6. From the two figures follows that attenuation at the ear (Fig. 6) is much smaller than at the error microphone (Fig. 5). The discrepancy is even more evident when considering attenuation of fundamental frequencies of the tones [7], [11].

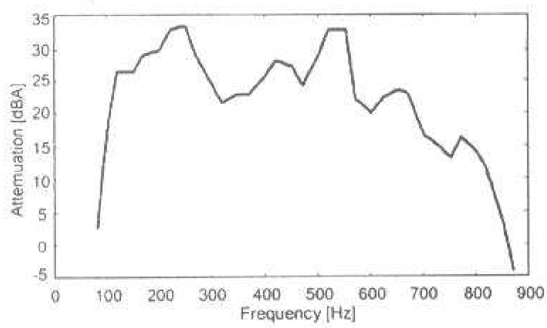

Fig. 5 Attenuation at the error microphone.

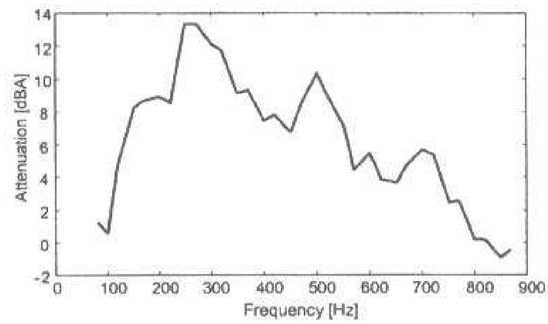

Fig. 6 Attenuation at the observer microphone.

Distribution of zones of quiet, obtained for tonal noise of frequency 250 Hz is presented in Fig. 7. The attenuation reported was measured with Brüel & Kjær 2235 sound level meter at about 300 points over a surface crossing the ears.

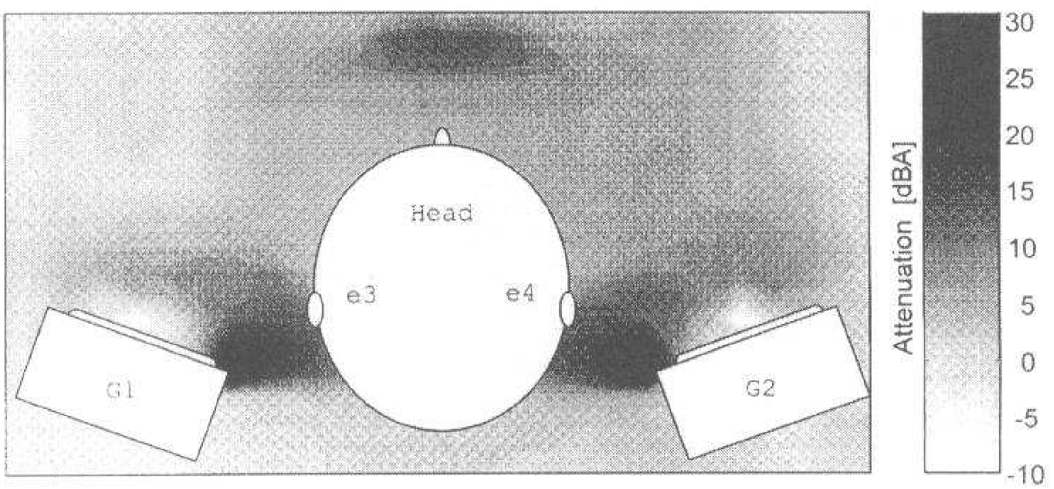

Fig. 7 Distribution of zones of quiet for the classical IMC control system.

This figure gives insight into shapes and dimensions of the attenuations areas. It confirms that they are irregular and the highest attenuation appears at the error microphones. At the diaphragms of loudspeakers noise reinforcement is observed. Although the highest possible attenuation reaches 30 dB the ears are caught by 12 dB zones only. Such attenuation is also achievable for practical lateral head movements. In case of forward movements, the attenuation significantly decreases and when passing the distance of 4 cm the ears fall into the 5 dB zone. This zone is of large dimension and therefore covers even substantial further position changes.

4 VIRTUAL MICROPHONE APPROACHES

From the experimental results presented hitherto follows that the generated zones of highest attenuations are generally of small dimensions and do not cover the human's ears in a practical system. It is thus justified to make an effort to shift them to desired locations, namely to the virtual (referred to so far as the observer) microphones as illustrated in Fig. 8 – see also [8-10].

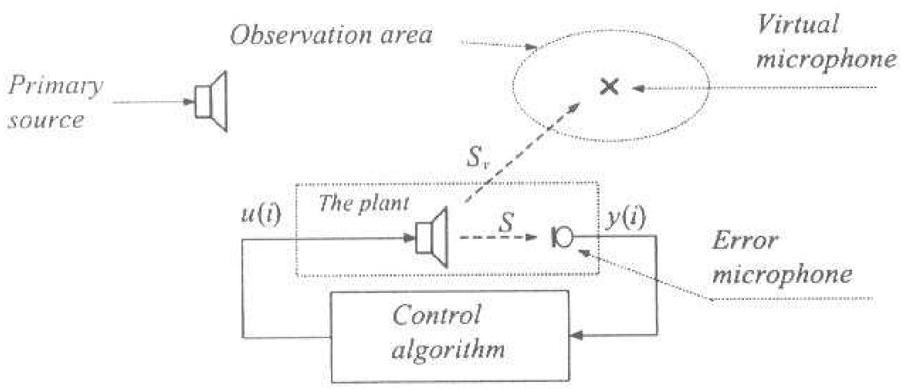

Fig. 8 The idea of virtual microphone.

Obviously, such microphones (e3 and e4 in Fig. 1) cannot be employed in a control system. In the following, two versions of adaptive IMC incorporating the idea of virtual microphone will be presented: designed for stationary and non-stationary noise.

4.1 STATIONARY NOISE CASE

Block diagrams of the proposed control system are presented in Fig. 9 and Fig. 10. In the tuning stage (see Fig. 9) it is necessary to locate temporarily, prior to experiments, an observer (the so-called virtual) microphone, e3 (e4), in the observation area – where the highest attenuation is required, i.e., at the ear. The control algorithm, FXLMS, minimises MSE of the signal $y_v(i)$ measured by that microphone. In the same time parameters of an additional filter H are estimated by another LMS algorithm that suppresses MSE of $y'(i)$. Signal $y'(i)$ is the difference between the output, $y(i)$, and signal $x'(i)$, i.e., estimated reference signal, $x(i)$, shaped by that filter. Therefore, filter H embeds knowledge about the output, $y(i)$, that exists while the highest attenuation appears at the virtual microphone.

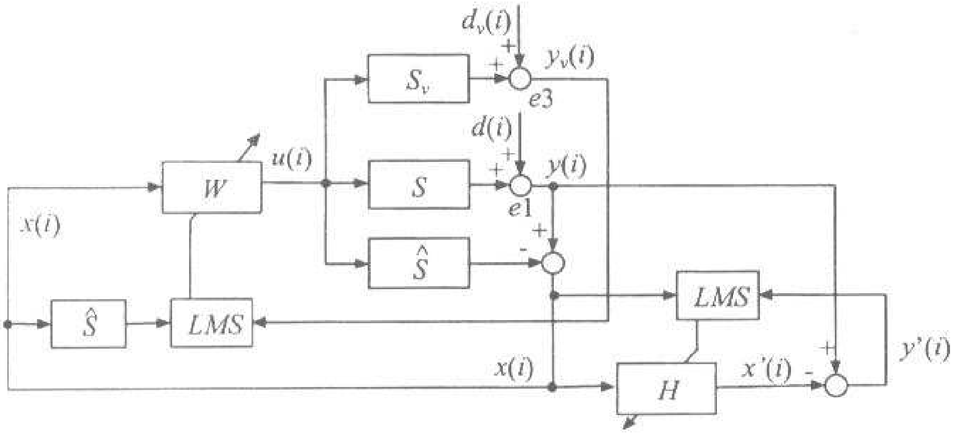

Fig. 9 Control system with virtual microphone for stationary noise – tuning stage.

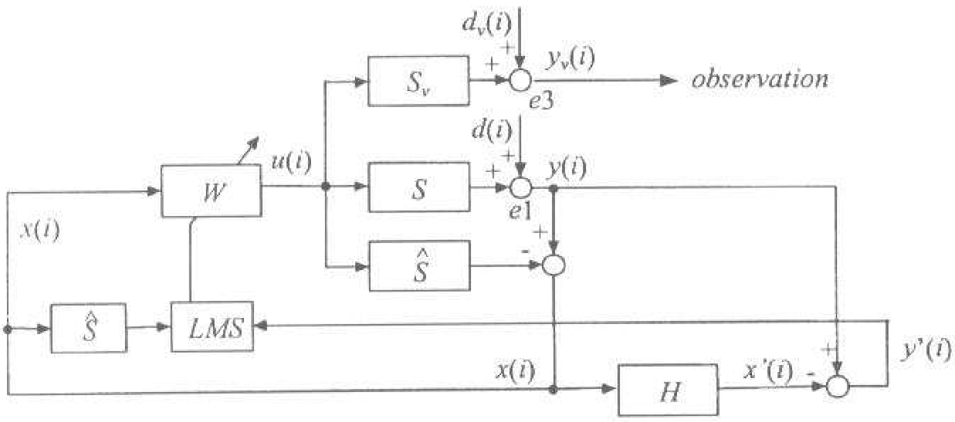

Fig. 10 Control system with virtual microphone for stationary noise – control stage.

In the control stage (see Fig. 10) the FXLMS algorithm uses, in turn, the signal $y'(i)$ but not directly $y(i)$ as it was in the classical system (Fig. 4) or $y_v(i)$ as it was in the tuning stage (Fig. 9). Since the FXLMS algorithm tends to minimise MSE of $y'(i)$, the signal $x'(i)$ can be interpreted as the set point for the output, $y(i)$ that implies the highest attenuation at e3, i.e., the ear. Similarly to the previous case this system was then implemented in multivariable version – see also [7], [11].

Distribution of the zones of quiet, obtained as previously, for a tone of 250 Hz is illustrated in Fig. 11. This figure confirms that the areas of highest attenuation are in desired locations. Their dimensions extended comparing to Fig. 7 and for 25 dB reached 7 cm. This is a consequence of the fact that presence of the head imposes zero pressure gradient at its surface, what "flattens" the secondary field and thereby extends the zones close to the head [8], [9]. Further head movements lead to leaving subsequent zones. However, the 17 dB zone covers the movements of about 15 cm to the left and 30 cm to the right. The non-symmetry is due to location of the primary source. Forward movements change the zones more rapidly, i.e., the attenuation drops down faster until reaching the 5 dB zone that extends over all practical head positions.

Further observations performed on the system with the aid of a spectral analyser demonstrate that practical changes in the plant do not influence performance of the entire system significantly, although the filter H is not updated. Thus, its parameters can be stored in memory allowing to omit the tuning stage.

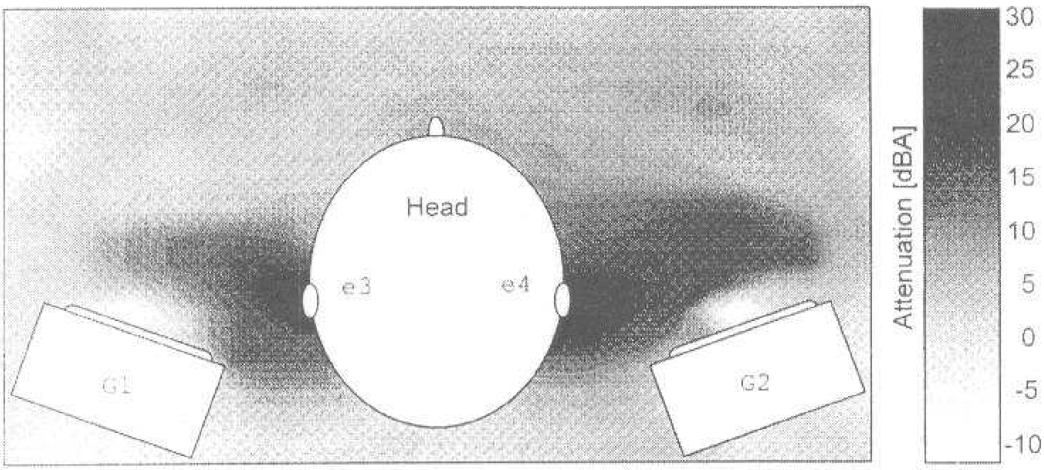

Fig. 11 Distribution of zones of quiet for the system with virtual microphone designed for stationary noise.

4.2 NON-STATIONARY NOISE CASE

The system presented above provides very high attenuation at desired locations. Unfortunately, its concept is based on the assumption that the noise is stationary to allow to tune and use filter H. In many real working environments this constraint is satisfied, e.g., the noise generated by an engine or fan operating with constant rotation rate. However, there are also many other devices where noise spectrum changes, e.g., in a car. Therefore, to cope with such noise another algorithm has been designed – see Fig. 12. It is based on the assumption that the primary noise source is far enough from the headrest for the difference between its contribution to field response at location of the error, $d(i)$, and the virtual, $d_v(i)$, microphones to be negligible, i.e., $d_v(i) \approx d(i)$. This is true provided spacing of these microphones is small compared to the acoustic wavelength [8]. In this article results for the tone of 250 Hz are reported. Its wavelength is about 136.0 cm, what is much less than the distance of the microphones being about 7.2 cm. Under this assumption the estimate of the acoustic disturbance at the error microphone can be considered as the estimate of the acoustic disturbance at the virtual microphone, $\hat{d}_v(i) \approx \hat{d}(i)$. Then, provided a model, $\hat{S}_v$, of the virtual path, S_v, is known, the residual signal at the virtual microphone can be estimated as

$$\hat{y}_v(i) = x(i) + \hat{\mathbf{s}}_v * \mathbf{u}(i), \tag{3}$$

where $\hat{\mathbf{s}}^T = [\hat{s}_{v0}, \hat{s}_{v1}, \ldots, \hat{s}_{v(L-1)}]$ is the vector of parameters of FIR model, $\hat{S}_v$. Referring to (1) it can be further expressed as

$$\hat{y}_v(i) = y(i) + \left(\hat{\mathbf{s}}_v - \hat{\mathbf{s}}\right) * \mathbf{u}(i). \tag{4}$$

Thus, the "quality" of $\hat{y}_v(i)$ estimate depends on modelling the difference between the virtual and secondary paths. Contribution of the secondary source to the outputs $y(i)$ and $y_v(i)$ cannot be assumed the same as it was for the primary source due to strong acoustic near field of the secondary loudspeaker and therefore (4) makes sense. The MSE of the estimated virtual microphone error signal, $y_v(i)$, can then be minimised with FXLMS algorithm (see Fig. 12).

This algorithm does not assume noise stationarity what means that it can be successfully applied for both stationary and non-stationary noise. However, for comparison of performance of all the presented algorithms, distribution of zones of quiet, obtained as previously, for the stationary tone of 250 Hz is presented in Fig. 13. It is seen that the areas of highest attenuation are well located although the attenuation is significantly lower than in the design for stationary noise. This can be explained by the fact that the assumption underlying this algorithm was not fully met. The conclusion that dimensions of the zones extend close to the head, explained in the previous section, still holds. Moreover, in this case the zones are larger and both lateral and forward head movements do not change the attenuation dramatically. For practical head positions attenuation higher than 8 dB is guaranteed.

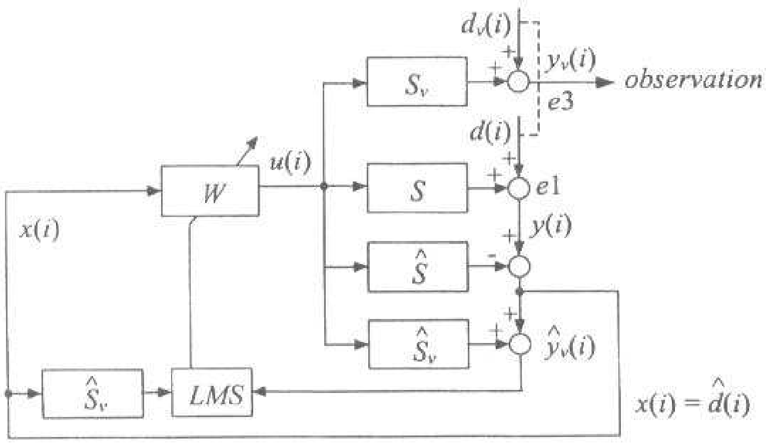

Fig. 12 Control system with virtual microphone for non-stationary noise.

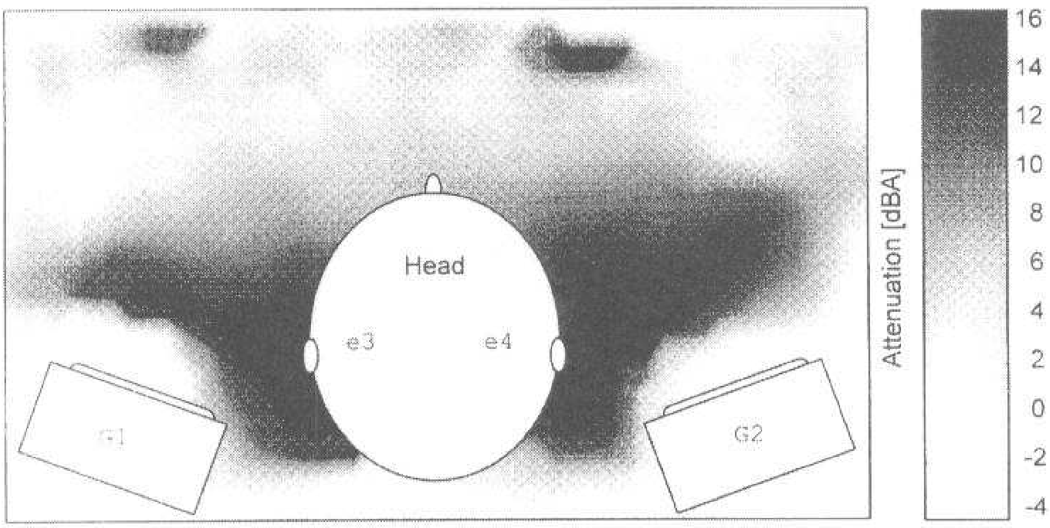

Fig. 13 Distribution of zones of quiet for the system with virtual microphone designed for non-stationary noise.

5. CONCLUSIONS

In this article an active headrest system has been considered. It constitutes a device that aims at providing acoustic comfort to the user sitting in a chair located in a noisy environment. The headrest system used for laboratory experiments has been described. It has been experimentally proved that classical IMC structure provides high attenuation but the zones of quiet are generated at the error microphones mounted at the headrest and do not reach the human's ears. Therefore, the idea of virtual microphones has been adopted. In the sequel, two extensions of the IMC system have been designed and practically verified. The first one guarantees attenuation at the ears similar to that achievable so far at the error microphones. However, this algorithm can be successfully used if the noise is stationary. To fill in the gap emerged another algorithm has been presented that can be applied for both stationary and non-stationary noise. Instead, to operate efficiently the primary source should be located sufficiently far from the headrest. The price paid for extending the system flexibility is degrading the performance in terms of the attenuation level. However, the subjective impression perceived by the listener is astonishing. As the attenuation and its gradient are lower its change in case of head movements is also smaller what less confuses the user. Obtained results have been presented graphically as distribution of attenuation areas.

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REFERENCES

1. S.J. Elliott, *Signal Processing for Active Control*, London, Academic Press, 2001.
2. S.M. Kuo, D. R. Morgan, *Active Noise Control Systems. Algorithms and DSP implementations*, New Jersey, J. Wiley & Sons, 1996.
3. M. Morari, E. Zafiriou, *Robust Process Control*, New Jersey, Prentice-Hall, 1989.
4. P.A. Nelson, S.J. Elliott, *Active Control of Sound*, Cambridge, Academic Press, 1994.
5. H.F. Olsen, E.G. May, "Electronic sound absorber", *Journal of the Acoustical Society of America* 25, 1130-1136, (1953).
6. M. Pawelczyk, "Analysis of active noise control plants using process identification and signal processing methodology". *Archives of Acoustics* 26(2), 143-164, (2001).
7. M. Pawelczyk, "Control of sound in an active headrest system", *Proceedings of Methods and Models in Automation and Robotics – MMAR 2002*, (2), 1157-1162, (2002).
8. B. Rafaely, S.J. Elliott, J. Garcia-Bonito, "Broadband performance of an active headrests", *Journal of the Acoustical Society of America* 106(2), 787-793, (1999).
9. B. Rafaely, S.J. Elliott, "H2/H-infinity active control of sound in a Headrest: design and implementation", *IEEE Transactions on Control Systems Technology* 7(1), 79-84, (1999).
10. W. K. Tseng, B. Rafaely, S.J. Elliott, "Performance and stability of single channel virtual microphone feedback controllers for an active headrest system", *Proceeding of International Conference on Optimisation and Optimal Control*, Taiwan, (2001).
11. M. Wilkosz, *Active noise control in headrests. M.Sc. Thesis (in Polish)*, Silesian University of Technology, Gliwice, Poland, 2001.