Application of the Remote Microphone Method to Active Noise Control in a Mobile Phone

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Mobile phones are used in a variety of situations where environmental noise may interfere with the ability of the near-end user to communicate with the far-end user. To overcome this problem, it might be possible to use active noise control technology to reduce the noise experienced by the near-end user. This paper initially demonstrates that when an active noise control system is used in a practical mobile phone configuration to minimise the noise measured by an error microphone mounted on the mobile phone, the attenuation achieved at the user's ear depends strongly on the position of the source generating the acoustic interference. To help overcome this problem, a remote microphone processing strategy is investigated that estimates the pressure at the user's ear from the pressure measured by the microphone on the mobile phone. Through an experimental implementation, it is demonstrated that this arrangement achieves a significant improvement in the attenuation measured at the ear of the user, compared to the standard active control strategy. The robustness of the active control system to changes in both the interfering sound field and the position of the mobile device relative to the ear of the user is also investigated experimentally.

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I. INTRODUCTION

Active noise control has now become a practicable technology and has been successfully employed in a variety of applications where it is not possible to achieve sufficient levels of control using passive noise control measures. For example, one of the most successful applications of active noise control technology is in headphones[1], where it is not possible to achieve high levels of low frequency noise control due to practical constraints on the size and weight of the headphones. Active noise cancelling headsets have been shown to achieve an additional 20 dB of noise attenuation up to 1 kHz[2]. More recently, active noise control has been extended to in-ear headphones, where the constraints on passive isolation are potentially even more restrictive[3]. Active noise control has also found application in the automotive[4–6], marine[7–9] and aerospace environments[10,11].

In the mobile phone, or more broadly mobile device application, active noise control has also been of recent interest[12] and this is due to the increasing expectations of mobile phone users in terms of both audio quality and functionality, and communication quality[13]. Mobile phones are often used in acoustic environments with high levels of background noise and, therefore, it is desirable to reduce the interference that this noise causes to the near-end user. This cannot generally be achieved using passive noise control treatments in the mobile phone application and, therefore, there has been commercial interest in implementing active noise control systems in mobile phones[12]. Although there has been limited published research into the practical design and limitations of mobile phone active noise control systems, Kottayi *et al* have recently investigated the effect of an active noise control system on speech intelligibility in a potential mobile phone application[14]. To provide insight into the practical limitations of applying an active noise control system to the mobile phone problem, this paper presents an investigation into the implementation and performance limitations of such a system.

In addition to investigating the performance limitations of a mobile phone based active noise control system using a standard feedforward active noise control strategy, the potential benefit of employing a remote microphone processing strategy is also investigated. In the mobile phone active noise control system, the physical error microphone is inevitably mounted on the body of the mobile phone and, therefore, it is not clear whether controlling the sound at this position will result in a reduction in the noise level at the ear of the user. A similar problem occurs in active headrest systems, where it is not possible to locate the error microphones in the ears of the user. To overcome this problem both virtual and remote microphone processing strategies have been proposed[15–17], which attempt to estimate the pressure at the desired, virtual, cancellation position using the pressure measured by a physical, monitoring error microphone. These processing strategies have been used to shift the zone of quiet generated by a local active noise control system to be targeted at the location of the user's ear[18] and such systems have recently been made adaptive to the position of the user's head by incorporating head tracking technology into the control system[19]. In this paper, the remote microphone method, originally proposed in[16] and investigated more recently in the context of spatially random pressure fields in[20], will be applied to the mobile phone active noise control system. In particular, the system employing the remote microphone method will be compared to the standard active noise control system in the context of the mobile phone application.

In the following section, the architecture of the mobile phone active noise control system is described and in Section III the control strategies are described, which include the feedforward filtered-reference least mean square (LMS) algorithm with and without the remote microphone strategy. In Section IV the implementation and real-time testing of the active noise control systems are described. In particular, the effect of changes in the primary disturbance sound field and changes in the position of the phone relative to the head of the user are investigated. Finally, in Section V conclusions are drawn.

II. MOBILE PHONE ANC ARCHITECTURE

Active noise control systems generally utilise loudspeakers as the control sources and microphones as the error sensors, whilst the type of reference sensors will depend on the application of the system. For example, in the automotive engine noise control problem, a reference signal is provided from a tachometer[4], since this provides a time-advanced reference signal that is directly correlated with the disturbance noise source. In the mobile phone application, since the disturbance noise will be generated by a variety of different sources, which will also change over time, a reference signal cannot generally be obtained directly from the disturbance noise source. Instead, a microphone mounted on the mobile phone must be used as the reference sensor, as illustrated in Figure 1. The ability of this reference sensor to provide a time-advanced reference signal to the causally constrained controller will be critical to the performance of the mobile phone feedforward ANC system. Figure 1 also shows an example of the positions of the loudspeaker used for control and of the monitoring error microphone on the mobile phone. The loudspeaker position is consistent with the standard location on the mobile phone and the monitoring error microphone position has been selected to be as close as possible to the expected position of the user's ear. Although a larger zone of quiet, and therefore a system that is more robust to changes in the user position could be achieved by using multiple control sources and error sensors[20,21], this has not been considered here due to the practical constraints on space and cost associated with integrating multiple transducers within a mobile phone.

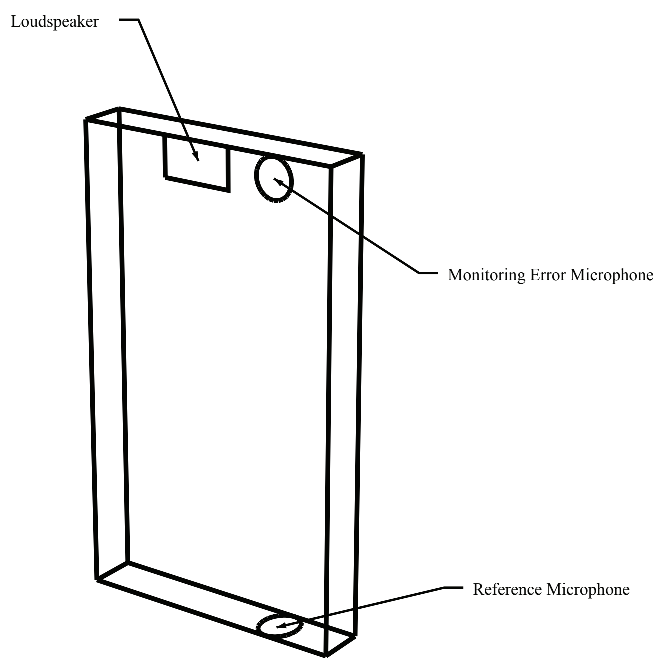


FIG. 1. Mobile phone geometry showing the locations of the monitoring error microphone, the reference microphone and the control loudspeaker.

III. ACTIVE CONTROL STRATEGIES

There are a wide variety of different control strategies that have been used to implement active noise control systems. These different strategies can be broadly separated into feedforward and feedback systems and the different approaches have been extensively reviewed in a number of textbooks on the subject[22–24]. When the control application requires adaptation due to changes in the disturbance over time, the filtered-reference LMS algorithm has probably been the most widely applied control strategy due to its simplicity of implementation and practical robustness. In the following section, the feedforward filtered-reference LMS algorithm will be briefly reviewed for the mobile phone control system and subsequently the modified control strategy integrating the remote microphone method will be presented.

A. Standard Feedfoward Control

The aim of the standard active noise control system is to minimise the pressure measured directly by the error microphone. In a feedforward control architecture, this is achieved by adaptively filtering a reference signal and driving the control loudspeaker to minimise the error signal. In the mobile phone system shown in Figure 1, the standard feedforward control system attempts to minimise the signal measured by the monitoring error microphone by adaptively filtering the signal measured by the reference microphone to drive the control loudspeaker, as shown in Figure 2. The broadband cost function to be minimised is defined in this case as the expectation, E, of the squared monitoring error signal,

$$J = E\left[e_m^2(n)\right], \tag{1}$$

where $e_m(n)$ is the monitoring error signal sampled at the n-th time step.

This cost function can be minimised using the filtered-x LMS algorithm, as shown in Figure 2. The derivation of this algorithm can be found in standard text books on active control, for example[24], and therefore will only be summarised here briefly. Using the filtered-x LMS algorithm, the vector of causally constrained control filter coefficients are updated as

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha \mathbf{r}_m(n) e_m(n), \tag{2}$$

where $\mathbf{w}$ is the vector of I control filter coefficients given by

$$\mathbf{w} = \begin{bmatrix} w_0 & w_1 & \cdots & w_{I-1} \end{bmatrix}^T, \tag{3}$$

$\mathbf{r}_m(n)$ is the vector of the current and past values of the filtered reference signal, which are given by filtering the reference signal, $x(n)$, by the plant response , $\mathbf{g}_m$ as

$$r_m(n) = \mathbf{g}_m^T \mathbf{x}(n), \tag{4}$$

where $\mathbf{g}_m$ is the vector of I_m Finite Impulse Response (FIR) filter coefficients that represent the plant response, and α is the step size or convergence gain.

In practice, the plant model used to calculate the filtered reference signal according to eq. (4) will be an estimation of the physical plant response, which can be represented by the vector of plant model filter coefficients, $\hat{\mathbf{g}}_m$. As a result, the filtered reference signals used in the practical update algorithm are also estimates and the practical filtered-x LMS algorithm is

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha \hat{\mathbf{r}}_m(n) e_m(n). \tag{5}$$

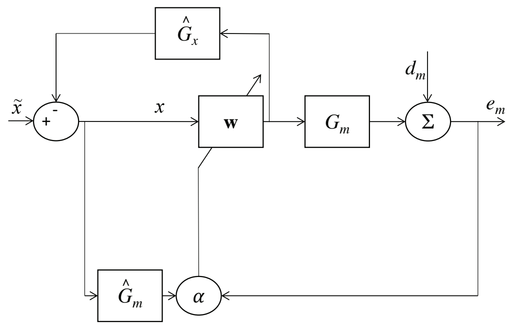


FIG. 2. Block diagram of the feedforward filtered-reference LMS algorithm with feedback path cancellation.

In addition, in the mobile application considered here, since the reference sensor is a microphone, there will be acoustic feedback from the control loudspeaker to this sensor and, therefore, this must be cancelled electronically to ensure that the controller operates as a feedforward system, as shown in Figure 2. The reference signal used in the control filter update equation is thus given by

$$x(n) = \tilde{x}(n) - \hat{\mathbf{g}}_x^T \mathbf{u}(n), \tag{6}$$

where $\hat{\mathbf{g}}_x$ is the vector of I_x FIR filter coefficients that is used to model the response between the control loudspeaker and the reference sensor and $\tilde{x}(n)$ is the signal measured by the reference microphone, which is affected by feedback from the controller.

B. Feedforward Control with Remote Sensing

In the mobile phone application, it is desirable to control the pressure at the user's ear rather than that measured by the monitoring microphone, which is mounted on the body of the device and, therefore, at some distance from the user's ear. The cost function in this case is given by

$$J = E\left[e_e^2(n)\right], \tag{7}$$

where $e_e(n)$ is the error signal measured at the user's ear. In practice, it is generally not possible to install an error microphone in the user's ear, as noted in the introduction, however, this problem has been solved in other applications by using the remote microphone method. The remote microphone method was originally proposed in[16] and can be used in the mobile phone application to estimate the pressure at the ear location from the pressure at the monitoring error microphone. This estimate of the error at the ear location, $\hat{e}_e(n)$, is given by

$$\hat{e}_e(n) = \hat{d}_e(n) + \hat{\mathbf{g}}_e^T \mathbf{u}(n), \tag{8}$$

where $\hat{d}_e(n)$ is an estimate of the disturbance signal that would be measured at an error microphone located in the ear and $\hat{\mathbf{g}}_e$ is a model of the plant response between the control loudspeaker and the ear error microphone. According to the remote microphone method, the disturbance at the ear can be estimated from the disturbance at the monitoring microphone via the linear observation filtering operation given by

$$\hat{d}_e(n) = \mathbf{o}^T \mathbf{d}_m(n) \tag{9}$$

where $\mathbf{o}$ is the vector of I_o observation filter coefficients and $\mathbf{d}_m(n)$ is the vector of the current and past values of the disturbance signal at the monitoring microphone, which can be estimated in real-time from the error signal measured at the monitoring microphone by cancelling the contribution from the control action as

$$\hat{d}_m(n) = e_m(n) - \hat{\mathbf{g}}_m^T \mathbf{u}(n). \tag{10}$$

The optimal value of the observation filter coefficients, $\mathbf{o}$, can be calculated by minimising the expectation of the squared error defined as the difference between the physical disturbance at the ear, $d_e(n)$, which can be measured directly as part of a preliminary controller calibration process, and the estimated disturbance at the ear, $\hat{d}_e(n)$. In practice it is necessary for this observation filter to be causally constrained and the optimal filter is given in this case by[20]

$$\mathbf{o}_{opt} = \mathbf{R}_{mm}^{-1} \mathbf{r}_{me}, \tag{11}$$

where

$$\mathbf{R}_{mm} = E\left[\mathbf{d}_m(n)\mathbf{d}_m^T(n)\right] \tag{12}$$

is the autocorrelation matrix corresponding to the disturbance signals at the monitoring microphone and

$$\mathbf{r}_{me} = E\left[\mathbf{d}_m(n) d_e(n)\right] \tag{13}$$

is the vector of cross correlations between the disturbances measured at the ear error microphone and the monitoring microphone.

Having estimated the error signal at the ear position using the remote microphone method described above, it is then possible to implement the filtered-x LMS algorithm as described in the previous section, but with a modification so that the controller minimises the estimated error signal at the ear location. In this case the practical version of the filtered-x LMS algorithm is given by

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \alpha \hat{\mathbf{r}}_e(n) \hat{e}_e(n), \tag{14}$$

where $\hat{\mathbf{r}}_e(n)$ is the vector of current and previous samples of the reference signal filtered in this case by a model of the plant response between the control loudspeaker and the ear error microphone, such that

$$\hat{r}_e(n) = \hat{\mathbf{g}}_e^T \mathbf{x}(n). \tag{15}$$

The full implementation of the feedforward controller, employing the remote microphone method is shown by the block diagram in Figure 3. This shows the estimation of the error signal at the ear microphone, described by eqs. (8), (9) and (10); the reference signal feedback cancellation, described by eq. (6); and the adaptive control filter implementation, which is adapted according to eq. (14).

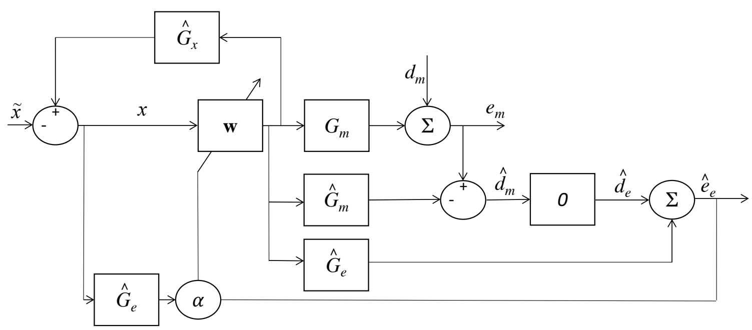


FIG. 3. Block diagram of the feedforward filtered-reference LMS algorithm with remote sensing.

IV. MOBILE PHONE ANC IMPLEMENTATION AND TESTING

To test the performance of the standard feedforward controller and the feedforward controller with remote sensing, the mobile phone mockup shown in Figure 1 has been constructed, as shown in Figure 4, and the two control algorithms have been implemented on

FIG. 4. Mockup of the mobile phone, with embedded loudspeaker, reference microphone and monitoring microphone (color online).

a rapid prototyping digital signal processing board. To test the control system, the mobile phone mockup has been installed in the anechoic chamber at the Institute of Sound and Vibration Research, along with a Kemar dummy head and 6 additional loudspeakers, as shown in Figure 5, which have been used to generate the disturbance, or primary sound field. The position of the loudspeakers used as primary sources relative to the dummy head and mobile device are also shown in Figure 6 with numbering to facilitate their identification in the following discussions.

In the first instance, the plant responses between the control loudspeakers and the monitoring, reference and ear microphones have been measured at a sample rate of $F_s = 10$ kHz, by driving the control loudspeaker with broadband white noise. Figure 7a shows the three

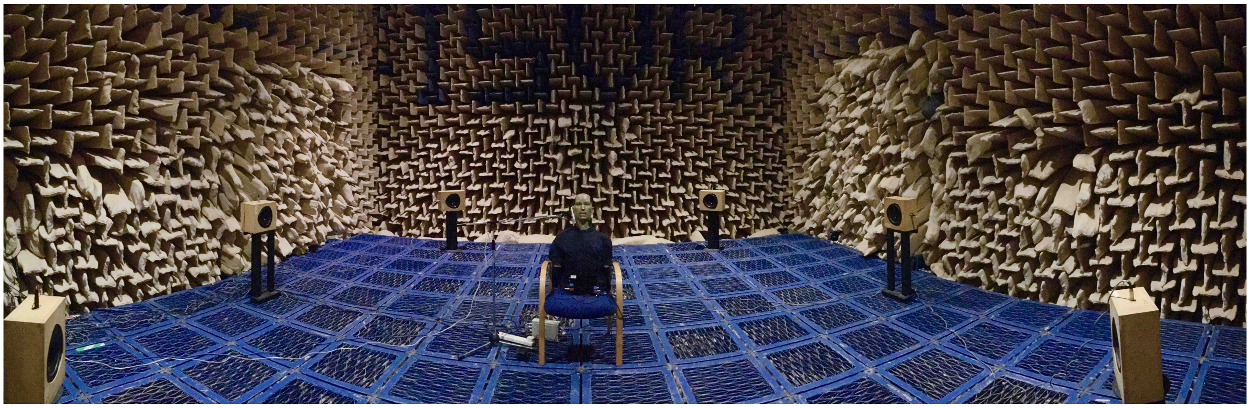


FIG. 5. Experimental setup in the ISVR anechoic chamber showing the Kemar dummy head and the six primary sources (color online).

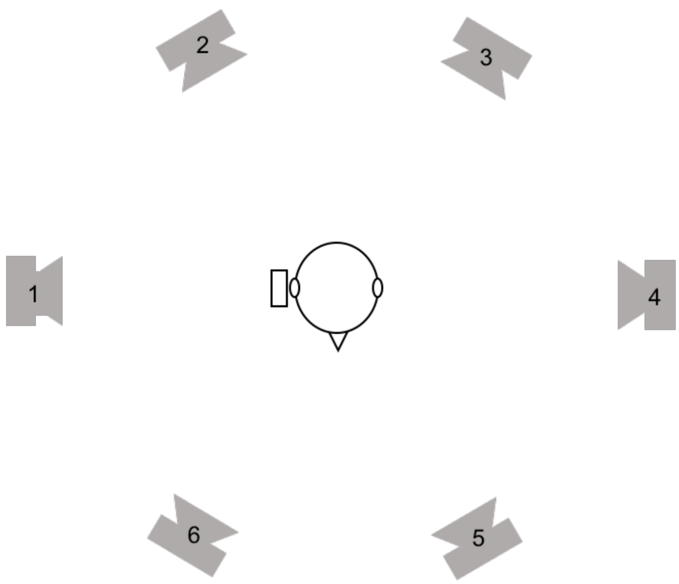
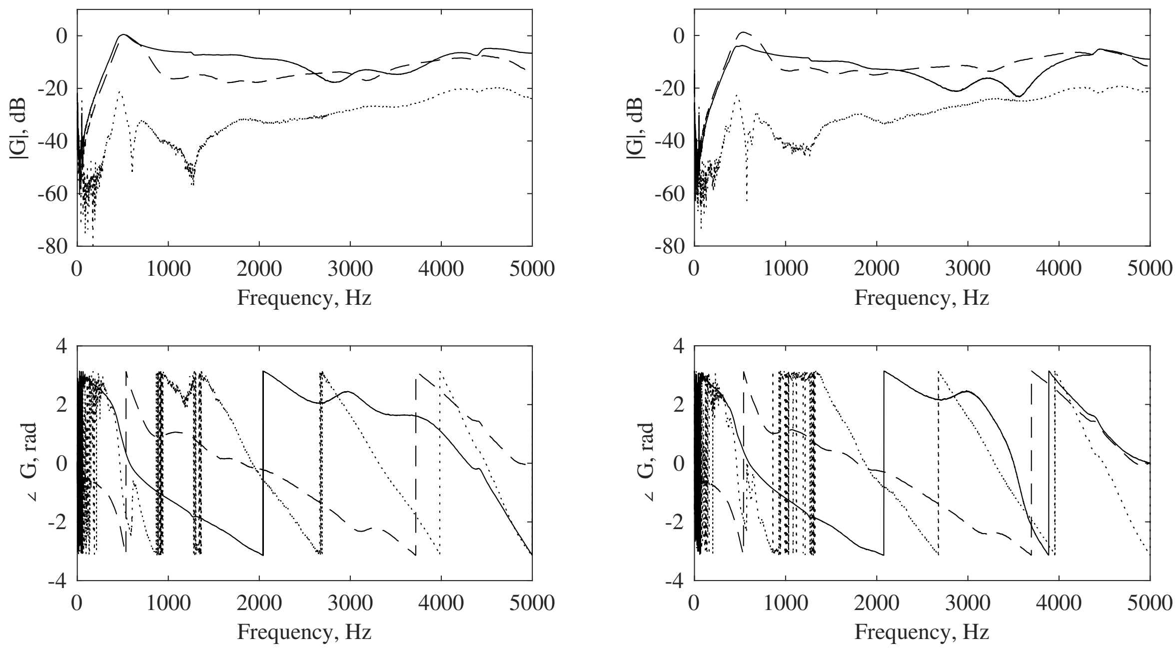


FIG. 6. Geometry of the primary sources used in the experimental testing shown in Figure 5.

frequency responses when the phone is located in the nominal position with respect to the user's head, whilst Figure 7b shows the same set of responses, but when the phone is moved a distance of about 3 cm away from the ear of the user. From these results it can be seen that the plant response to the monitoring error microphone is significantly modified by the change in phone position and, in particular, at frequencies above around 3 kHz the change in the position of the phone has introduced a 150° phase difference. The plant response between the control loudspeaker and the ear microphone in the dummy head shows a similar variation, although the difference in phase at higher frequencies is limited to 60° in this case. Finally, it can be seen that the response between the control loudspeaker and the reference microphone, which is mounted on the base of the phone, is significantly lower than to the other microphones, but it does not appear to be as significantly affected by changes in position as the error microphones.

A. Nominal Performance

In the first instance, the performance of the standard feedforward control system, described in Section III A, and the feedforward control system employing the remote microphone method, described in Section III B, have been assessed by simulation using measured responses under nominal operating conditions; that is, where the plant models used in the controllers are equal to the physical plant responses and where the optimal observation filter has been calculated according to eq. (11) for the actual disturbance noise conditions. The plant responses, G_m and G_e, and the feedback cancellation filter, G_x, were modelled using FIR filters with 128 coefficients. The control filter, $\mathbf{w}$, has been implemented with $I = 256$ coefficients and the observation filter, $\mathbf{o}$, has been implemented with 128 coefficients. In each case, the convergence gain, α, has been set at one half of the maximum stable value,



(a) (b)

FIG. 7. The frequency responses between the control loudspeaker and the monitoring error microphone (solid lines), the reference microphone (dashed lines) and the dummy head ear microphone (dotted lines) for the nominal phone position (a) and the perturbed position (b), where the phone is moved about 3 cm away from the ear.

which gives close to the fastest convergence speed, and the performance has been measured after the controller has adapted for 10 seconds.

Figures 8a and 8b show the performance of the two controllers at the ear microphone in the dummy head and the monitoring error microphone respectively when the primary disturbance sound field is generated by driving primary source 4, shown in Figure 6, with white noise. From the dashed line in Figure 8b it can be seen that the standard controller achieves a high level of reduction in the signal measured at the monitoring error microphone over the full bandwidth, however, from the dashed line in Figure 8a it can be seen that this does not translate into comparable reductions at the ear. Specifically, at higher frequencies, where the acoustic wavelength is short and thus the zone of quiet is limited in size, the attenuation at the ear location is limited. Conversely, it can be seen from the dotted line in Figure 8a that the controller employing the remote microphone method achieves a significant reduction in the signal measured at the ear location, whilst the attenuation at the monitoring location is limited in this case. These results demonstrate the potential performance advantage of employing the remote microphone method in the mobile phone application, however, it is important to understand how the controller performs for different primary disturbance sound fields.

Figures 8c and 8d show the performance of the two different controllers when the primary disturbance is generated by driving source 1, shown in Figure 6, with white noise. From these results it can be seen that the attenuation achieved by both controllers at both the monitoring and ear error microphones is limited compared to when the primary field is generated by source 4. Nevertheless, it is important to highlight that the controller employing the remote microphone method achieves a broadband attenuation in the error signal measured at the ear that is 2 dB greater than that achieved by the standard controller in this case and the peak reduction at the ear is 4 dB greater when using the remote microphone method.

The difference in the control performance achieved when the primary sound field is generated by either source 1 or source 4 can be related to the difference in the positions of these two primary sources with respect to the mobile device and the Kemar dummy head, as shown in Figure 6. In the case of primary source 4, an additional delay is thought to be introduced

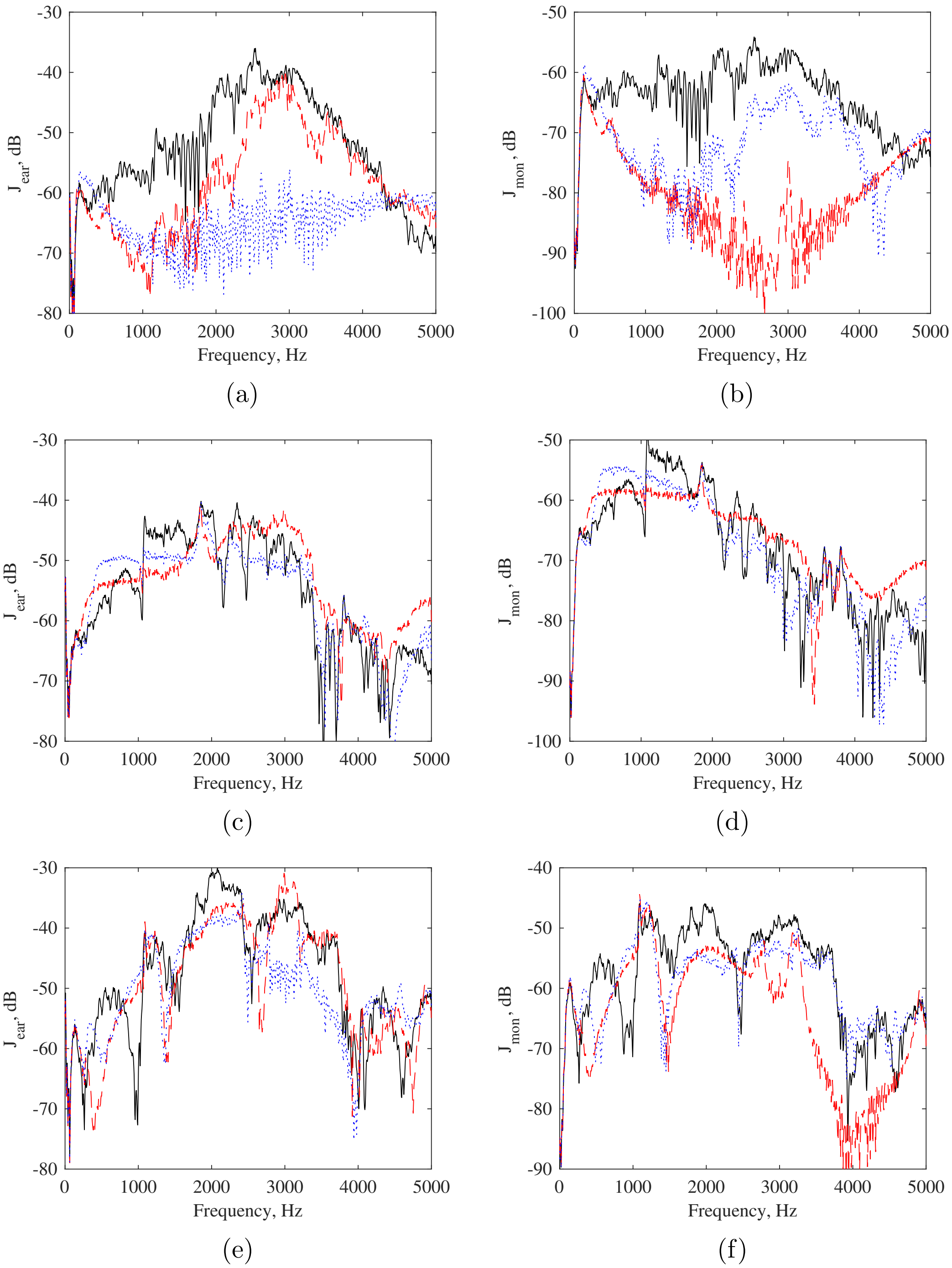


FIG. 8. The power spectral densities of the pressures measured at the ear microphone location (a, c, e) and at the monitoring microphone location (b, d, f) without control (black solid line), with standard feedforward control (red dashed line) and with feedforward control using the remote microphone method (blue dotted line) for primary sound fields generated by primary source 4 alone (a, b), primary source 1 alone (c, d) and all 6 primary sources driven with uncorrelated noise (e, f). (color online)

into the error signal paths due to the shielding provided by the diffraction of the primary sound field by the Kemar dummy head, which means that the reference signal benefits from a time-advance relative to the error signals. In the case of primary source 1, there is a direct path from the primary source to the reference microphone and both of the error microphones and, therefore, the head does not provide an additional time-advance. The presence of this additional time-advance can be verified by observing the cross-correlation between the reference and monitoring error microphones for primary source positions 1 and 4, which are shown in Figure 9. From the cross correlations it can be seen that for primary source 1, there is no time-advance provided by the reference microphone, whereas for source 4 there is a time-advance of 0.3 ms. These observations support the hypothesis that the additional control performance achieved when the primary sound field is generated by source 4 is due to the time-advance of the reference signal relative to the error signal. However, as noted in the similar discussion presented in[25] with reference to an ANC headphones implementation, it is also important to relate this time-advance to the delay in the plant response. For real-time feedforward active control, causality must be maintained and this means that the reference signal time-advance must be greater than the delay in the plant response, measured between the control loudspeaker and error sensor. In the mobile device system considered here, the delay in the plant response is 0.2 ms and, therefore, the time-advance in the case of primary source 4 is sufficient.

Finally, Figures 8e and 8f show the performance of the two control strategies at the ear and monitoring error microphones when the primary sound field is generated by driving all 6 primary sources, shown in Figure 6, with uncorrelated white noise. This disturbance

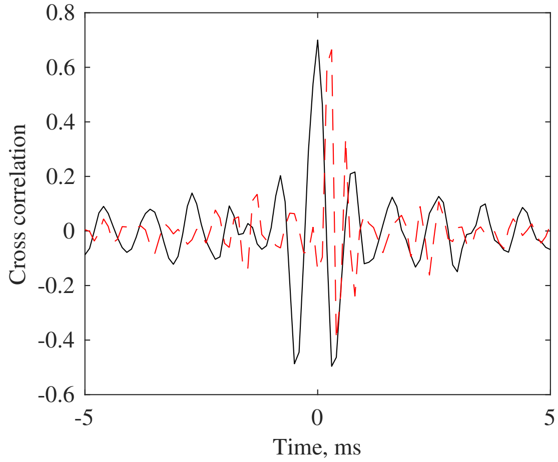


FIG. 9. The cross correlation measured between the reference microphone and the monitoring microphone for a primary sound field generated by source 1 (solid black) and source 4 (dashed red). (color online)

sound field may be closer to some practical scenarios, where the unwanted noise originates from multiple sources. From Figure 8f it can be seen that the standard controller achieves a higher level of attenuation at the monitoring microphone position than the controller using the remote microphone method, however, from the dotted line in Figure 8e it can be seen that the controller using the remote microphone method achieves a 3 dB broadband increase in the attenuation at the ear position compared to the standard controller. This again demonstrates the potential advantage for the user of employing the remote microphone method in the mobile device, but emphasises the variability in the performance due to variations in the primary sound field.

B. Robustness of the performance to changes in the Primary Disturbance

Although the results shown in Figure 8 have demonstrated the potential advantage of employing the remote microphone method in the mobile phone active noise control system, it is important to consider potential uncertainties that would be experienced in a practical implementation. These uncertainties have not been widely considered in the literature where the remote microphone method has been utilised. In particular, for example, in practice it may not be straightforward to update the observation filter, calculated according to (11), when there is a change in the primary sound field. Figure 10 shows the frequency responses of the optimal observation filters calculated for the three primary sound fields being considered here. From these responses it can be seen that there are significant differences between the optimal filters for the different sound field conditions in terms of both the magnitude and phase responses across the full frequency range. Therefore, it is important to consider how the performance of the controller employing the remote microphone method is affected when a fixed observation filter is utilised for different primary disturbance conditions.

Figure 11 shows the performance of the standard controller and the control system employing the remote microphone method when the observation filter has been calculated for a primary disturbance generated by driving all 6 primary sources with uncorrelated white noise. Figures 11a and 11b show the performance at the ear and the monitoring error microphones for the two controllers when the primary disturbance is generated by driving source 4 with white noise alone. From Figure 11a it can be seen that at the ear location the two controllers achieve different levels of attenuation in different frequency bands. The broad-

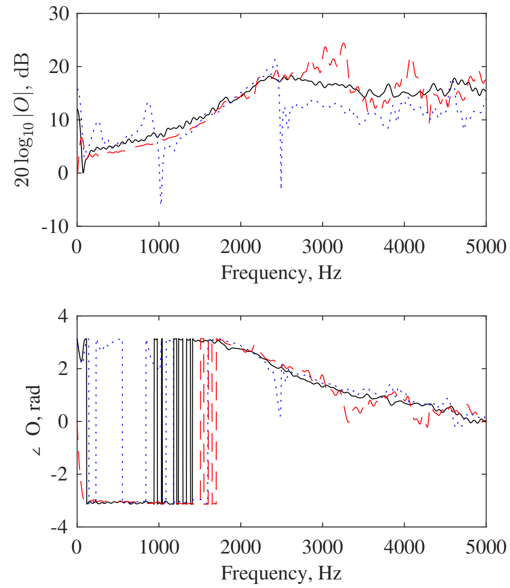


FIG. 10. Frequency responses of the optimal observation filters for a disturbance sound field generated by primary source 1 (red dashed line), primary source 4 (black solid line) and all 6 primary source (blue dotted line). (color online)

band average attenuation achieved by the standard controller is 5 dB, whilst the broadband attenuation achieved by the controller using the remote microphone method is 6 dB. The controller using the remote microphone method does, however, achieve a more significant improvement compared to the standard controller at frequencies above around 2.5 kHz, where the standard controller struggles due to the size of the zone of quiet. Figures 11c and 11d show the performance of the two controllers when the sound field is generated by driving primary source 1 with white noise, while the observation filter is still calculated for the condition when all 6 primary sources are driven with uncorrelated white noise. In this case the standard controller produces a broadband average *enhancement* of 2 dB, whilst the controller employing the remote microphone method maintains the broadband average level, i.e. a broadband attenuation of 0 dB. The controller using the remote microphone method, however, does achieve a peak attenuation of 12 dB, compared to 9 dB for the standard controller.

?From the results presented in Figure 11, where there is a change in the primary sound field but no change in the observation filter, it is clear that although the remote microphone controller does outperform the standard controller, in terms of the broadband attenuation achieved at the ear of the user, the performance is somewhat degraded compared to the remote controller with optimal observation filters. The effect that changes in the primary sound field have on the performance of the remote microphone method has not been previously studied and, therefore, these results highlight some of the limitations of the remote microphone method. They also highlight the need for an improved virtual sensing strategy, that can update the observation filter when the primary sound field changes. However, this is not possible using the present formulation of the remote microphone method because, as noted in Section III B, it requires prior knowledge of the pressure at the user's ear. An alternative strategy could potentially utilise a microphone array to estimate the position of the primary source and hence adapt the observation filter to changes in the primary field or use machine learning methods to adapt the observation filter to the primary disturbance field in real-time, but this would require a large set of training data to be obtained from real-world measurements.

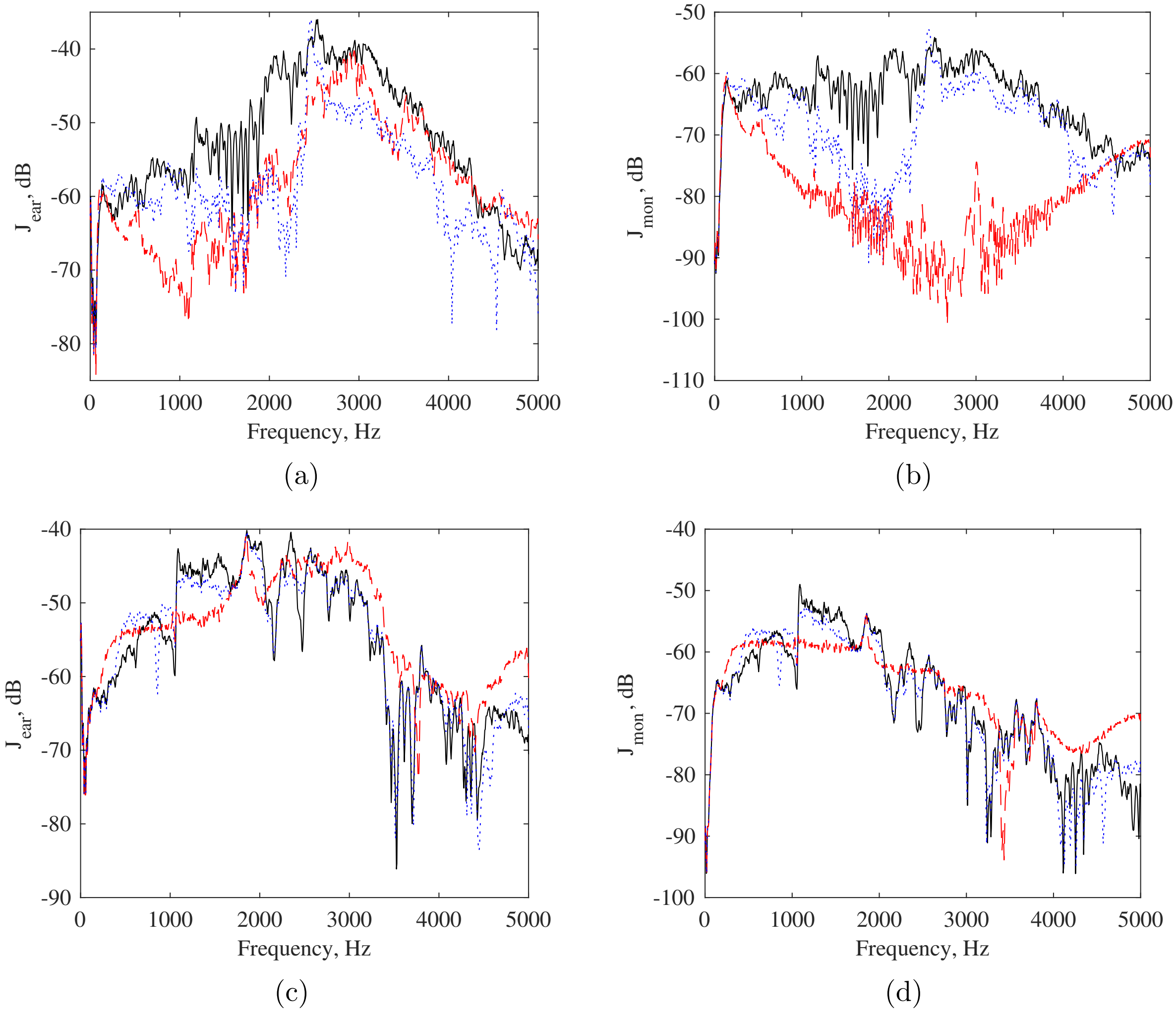


FIG. 11. The power spectral densities of the pressures measured at the ear location (a, c) and at the monitoring microphone location (b, d) without control (black solid line), with standard feedforward control (red dashed line) and with feedforward control using the remote microphone method when the observation filter has been designed based on the disturbance sound field generated when all 6 sources are driven with uncorrelated noise (blue dotted line) for primary sound fields generated by primary source 4 alone (a, b) and primary source 1 alone (c, d). (color online)

C. Robustness to Variations in the Phone Position

In addition to changes in the primary disturbance, in practice the mobile phone position will also change relative to the user's ear. Therefore, the robustness of the two control strategies to changes in the position of the device relative to the user's ear have also been investigated. A change in the position of the device relative to the user's ear will lead to a difference between the plant models used in the controllers ($\hat{\mathbf{g}}_m$, $\hat{\mathbf{g}}_e$ and $\hat{\mathbf{g}}_x$) and the physical responses. In addition, there will also be some modification in the disturbance signal and, therefore, the observation filter will no longer be optimal, as discussed above. However, it has been found that a change in the phone position has a much smaller influence on the optimal observation filters than changes in the primary sound field. Figure 12 shows the effect of changes in the phone position on the performance of the two controllers in terms of the signals measured at both the ear and the monitoring error microphones. Figures 12a and 12b show the results at the ear and monitoring microphones respectively for the standard controller. From Figure 12b it can be seen that as the phone is moved from the nominal position the attenuation achieved at the monitoring microphone is reduced. Interestingly, from Figure 12a it can be seen that for the standard controller, when the phone is moved closer the attenuation at the ear position is increased. This can be related to the closer proximity of the monitoring microphone, at which cancellation is focused in the standard controller, to the ear microphone.

Figures 12c and 12d show the corresponding results for the controller using the remote microphone method. From Figure 12d it can be seen that the attenuation achieved at the

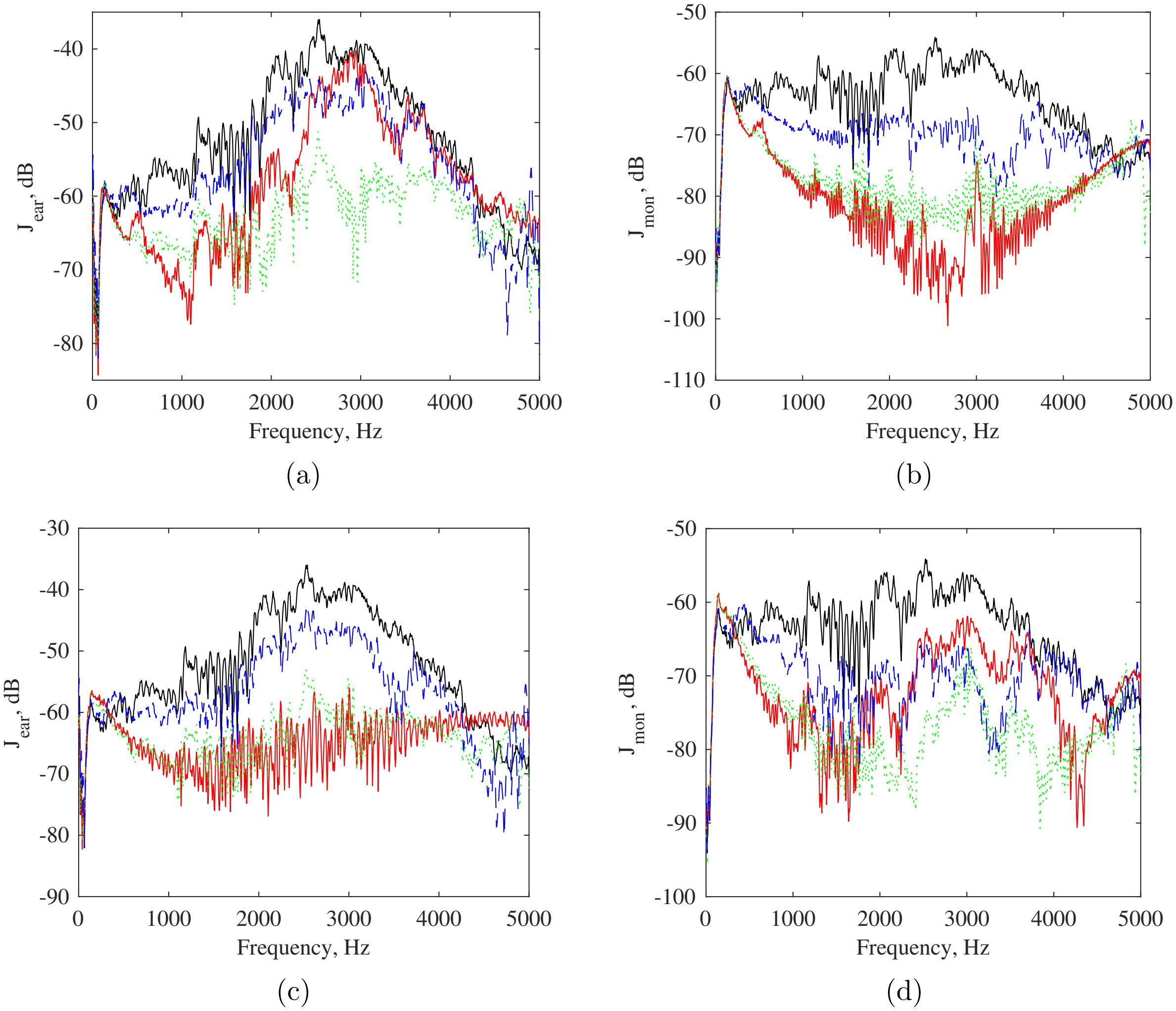


FIG. 12. The power spectral densities of the pressures measured at the ear location (a, c) and at the monitoring microphone location (b, d) without control (black solid line), with standard feedforward control (a, b) and with feedforward control using the remote microphone method (c, d) for a primary sound field generated by primary source 4, alone when the control system is designed using responses measured in the nominal position and when the phone is at the nominal position (red solid line), or at a greater distance from the ear (blue dashed line) or at a shorter distance from the ear (green dotted line). (color online)

monitoring microphone is relatively insensitive to changes in the position of the phone. However, from Figure 12c it can be seen that larger variations in the level after control are introduced at the ear position. Although the performance is not significantly modified when the phone is moved closer to the user's ear, when the phone is moved away from the user's ear the attenuation is degraded due to the additional delays introduced into the physical plant response compared to the modelled plant response leading to a slower convergence speed. To overcome this sensitivity it would be beneficial to update the plant response models used in the controller based on the position of the phone relative to the ear, as has been demonstrated in the active headrest application[19]. However, this would require some form of tracking technology, which is not straightforward in this context.

V. CONCLUSIONS

This paper has presented an investigation into the application of the remote microphone processing strategy in a mobile phone-based feedforward active noise control system. The aim of the remote microphone based control system is to improve the noise attenuation achieved at the user's ear compared to a standard active noise control system, which minimises the pressure at an error microphone mounted on the mobile phone rather than at the user's ear directly.

In the first instance, the performance of the two controllers has been investigated in terms of the pressures before and after control at both the on-phone monitoring error microphone and at the ear position, for different primary disturbance sound fields. Perfect knowledge of both the plant responses and optimal observation filters is initially assumed. Under these conditions, it has been shown that the performance achieved by both control strategies is strongly dependent upon the direction of arrival of the primary sound field. This is because the time-advance provided by the reference microphone mounted on the mobile phone depends on the location of the primary source relative to both the mobile phone and the head. Nevertheless, it has been shown that, when the reference signal provides sufficient time-advance, both systems are able to achieve significant levels of control at the error sensor, but that the remote microphone method is able to provide improved control at the ear position.

In practice there are likely to be uncertainties in both the plant responses and the knowledge of the primary sound field and, therefore, the influence of such uncertainties on the performance of the system have also been investigated. The effect of such uncertainties on the performance of an active noise control system utilising the remote microphone method have not previously been investigated. Therefore, in the first instance the observation filter used in the remote microphone processing strategy has been calculated for a known primary disturbance field, generated by multiple primary sources, and then the performance of the controller using this observation filter has been assessed when the primary sound field is modified. Under these conditions it has been shown that the performance advantage of the remote microphone strategy is significantly reduced and the broadband performance achieved by the system employing the remote microphone method becomes similar to that achieved by the standard controller. These results demonstrate the need for a more advanced remote microphone processing strategy, which is able to detect the features of the primary sound field and update the observation filter accordingly. It is not clear how this limitation can be overcome at this point, but it may require both a more advanced signal processing strategy, as well as a higher-order microphone array to detect the spatial properties of the primary sound field. For example, use could potentially be made of machine learning methods to develop an observation filter that can adapt to changes in the primary sound field.

Finally, the performance of the two control systems has been assessed when there is a change in the position of the mobile phone relative to the user's head. This results in a change in the plant responses from the secondary source to both the monitoring and remote microphone positions. From these results it has been shown that both controllers remain stable and continue to provide attenuation, for the perturbations investigated, but that the control performance is reduced in each case. For the control system employing the remote microphone method it has been found that the attenuation measured at the ear position is only slightly reduced when the phone is moved closer to the ear compared to the nominal position, but is significantly degraded when the phone is moved further away from the ear. In order to achieve the maximum level of control performance it is thus necessary to update the plant response models used in the controller based on the position of the phone relative to the ear. It may be possible that this could be achieved using some form of image-based tracking to determine the location of the phone relative to the user's ear.

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