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Adaptive design of a unidirectional source in a duct

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Abstract

An adaptive signal processing technique for the design of a unidirectional source for duct-acoustic applications is introduced. The source consists of multiple actuators. In this paper, two-element sources are considered. Directivity of the source is obtained by filtering the input signal of each actuator in an appropriate manner. The proposed adaptive system designs these filters automatically. One advantage of the approach is that the delay between the actuators—which is one of the main design parameters—need not be measured separately, since the system automatically learns the delay as part of the adaptation process. In addition, it becomes unnecessary to equalize the frequency responses of the actuators, since the adaptive system also accounts for this problem. Simulated examples demonstrate the effectiveness of the approach. The proposed technique may be used in an active noise control system for ventilation ducts, for example.

1. Introduction

An acoustic source consisting of multiple actuators may be designed so that it radiates essentially in one direction in a narrow duct [1–7]. The input signal is processed in an appropriate manner for each actuator. As a result, the actuator array radiates efficiently in one direction but in the other the output signals of the actuators cancel each other. The motivation for the design of such a unidirectional source is that it can be used in the implementation of a *feedforward* broadband active noise control (ANC) system in ducts, where the anti-noise needs to radiate downstream but it is at the same time desirable that it does not radiate upstream.

1.1 Advantages and disadvantages of unidirectional systems

The advantages of a unidirectional source in ANC systems are listed in the following [1], [3]:

1. The acoustic feedback between the secondary source and reference detector is eliminated, which stabilizes the generation of anti-noise using an adaptive system;
2. A feedback neutralization filter (acoustic echo canceller) is not needed, if the unidirectional source works properly, thus eliminating a possible source of instability in the control system;
3. The sound pressure level does not increase in the upstream direction due to the secondary source;
4. The sound pressure level may be attenuated in the upstream direction, since the unidirectional secondary source effectively absorbs the incident sound wave thus eliminating further reflections (from the open end of the duct, for example).

The main disadvantage of a unidirectional source is its limited frequency band [3]. Systems of multiple actuators cannot be unidirectional at very low frequencies (close to 0 Hz) and they have a principal upper frequency limit [7]. In practice the frequency band of unidirectional operation is 2 to 4 octaves, which is enough for many applications. Another disadvantage is naturally the need for several actuators, which increases the price of the system.

As a justification for unidirectional systems it may be pointed out that all digital ANC systems naturally have a limited frequency band (determined by the sampling frequency and the anti-aliasing filter). Also, multiple loudspeakers are often used in ANC systems in ducts for other reasons, for example to attenuate the propagation of higher-order modes. Thus it seems that these commonly highlighted disadvantages of unidirectional ANC systems are not particularly burdening.

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1.2 Motivation and background for the present work

The deviations in the frequency response of the actuators degrade the obtainable attenuation of unidirectional ANC systems. The mutual differences in the frequency responses are particularly harmful. For example, for a 20 dB difference in radiation efficiency to be obtainable between the down and upstream directions at a given frequency, the amplitude responses of the actuators should be identical within 1 dB and the phase responses with the accuracy of 5°.

The differences in the magnitude and phase response of loudspeakers may well exceed the above limits, and thus the mutual differences may be even larger. A clear consequence is that the digital filtering in a unidirectional ANC system should be designed so that it equalizes or compensates for the frequency response differences between the actuators and at the same time processes the input signal of the actuators so that the unidirectionality is achieved.

A measurement error in the distance of the acoustical centers of the actuators also degrades the unidirectional source, since it is one of the main parameters in the filter design for the unidirectional system. It has been shown that this error mainly affects the upstream radiation from the source [3]. Nevertheless, it would naturally be desirable to minimize this problem.

This paper proposes an adaptive approach to the design of digital filters for a unidirectional source. The weighting filters for the input signal of the actuators are designed using the multichannel filtered-x LMS algorithm [6].

The adaptive design approach is proposed as an alternative to the off-line design of the control system. The use of the proposed adaptive design approach makes it unnecessary to do extra measurements of the delay between the actuator elements or to equalize the frequency responses of the actuators, since these equalization and design steps are automatically accounted for. This renders the ANC system more reliable, robust, and easier to use. However, in the configuration to be presented the adaptation of the transfer functions between the loudspeakers and microphones needs to be done separately. The development of an adaptive system that takes care of all adaptation stages at the same time is left for future work.

In this paper, two-element unidirectional sources are considered. Section 2 discusses the formerly presented unidirectional sources constructed of two actuator elements. The new adaptive structure with some variants is introduced in Section 3, and simulation results that illustrate the effectiveness of the proposed technique are presented in Section 4. Section 5 concludes the paper and discusses possible directions for further research in this field.

2. Principal unidirectional two-element constructions

In the following, the two-element Swinbanks solution [1] and three JMC-based two-element solutions of Uosukainen and Välimäki [7] are explicated as the physical basis of this paper. All of them are ideal unidirectional solutions, giving the same outputs, but the realization structures for signal processing are different. The off-line design of these systems has been discussed in Ref. [7].

2.1 The Swinbanks solution

Swinbanks [1] has described an ideal two-element unidirectional solution where the upstream radiation is eliminated by introducing a delay τ for the first actuator, corresponding to the acoustic propagation delay between the two actuators, and feeding the actuators in opposite phases. The first actuator is situated upstream from the second one, their mutual distance being d. The control system of the Swinbanks source is shown in Figure 1.

The amplitude factor A in the control system is

$$A = \frac{kd}{\sin(kd)} \tag{1}$$

where k is the wavenumber. The delay τ_L corresponds to the propagation delay between the reference detector and the middle point of the actuator system, and c_0 is the speed of sound. The output of the first actuator is denoted by q_1 and that of the second one by q_2. The reference signal is denoted by q_L. The reference signal and the outputs of the loudspeakers are assumed to be of the same sort; no unit transformations are present in Figure 1. That is the case also in later figures of this paper. We also assume everywhere in this paper that the individual actuator elements are of monopole type.

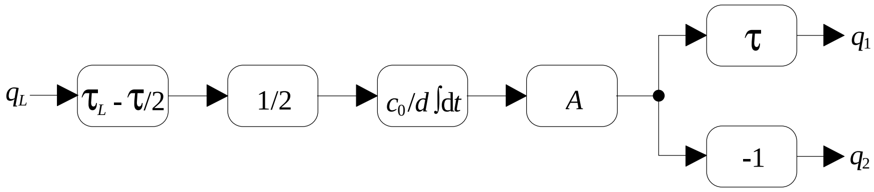

Figure 1. Control system of the two-element Swinbanks source (adopted from [7]).

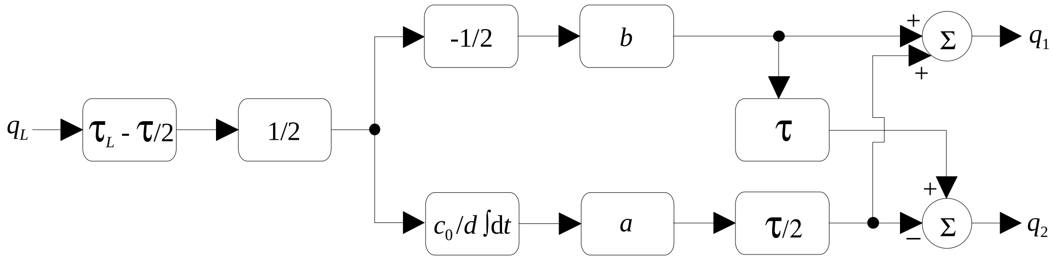

Figure 2. Control system of a two-element source with inter-channel delay optimized downstream [7].

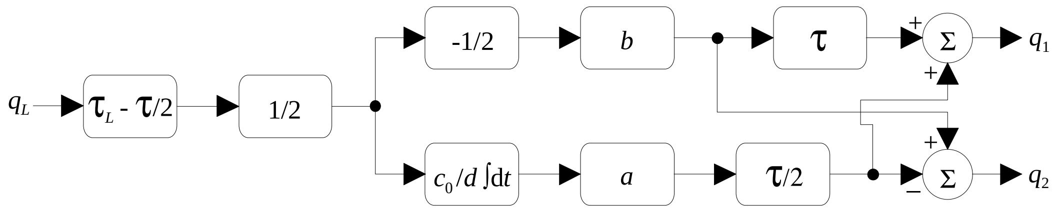

Figure 3. Control system of a two-element source with inter-channel delay optimized upstream [7].

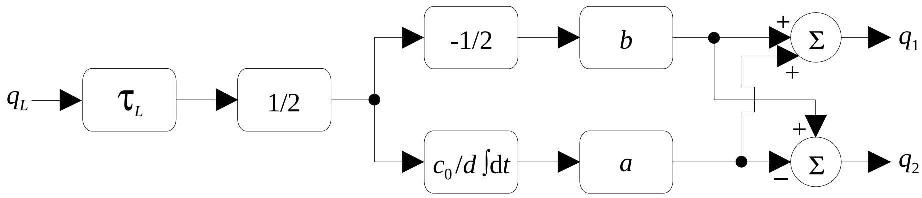

Figure 4. Control system of a two-element source with no inter-channel delay [7].

2.2 JMC-based solutions

The *JMC method* is suitable for formulating the problem of active noise control with the general system theory, although the method can be applied more generally to reshaping of acoustic fields [10–15]. Its name originates from the first three pioneers of the approach: Jessel, Mangiante, and Canévet [12]. Generally, three types of secondary sources are needed in the method—namely monopoles, dipoles, and quadripoles. In the case of duct applications (in the plane wave mode), the quadripoles vanish. The ideal actuator unit thus consists of an ideal monopole and an ideal dipole.

The inter-channel delay, corresponding to the acoustic propagation delay between the actuator elements, can be selected in the monopole part of the output q_2 in respect to that of the output q_1 to ensure that the monopole part of the radiation is correct downstream [7]. In that case, the control system feeding the actuators is as shown in Figure 2.

The inter-channel delay can be selected also to ensure that the monopole part of the radiation is correct upstream [7]. Figure 3 shows the control system for this choice.

One more choice is the one where there are *no delays* in the monopole parts ($\tau = 0$). For this case, the control system feeding the actuators is presented in Figure 4 [7].

Table 1 contains the values of weightings a and b for the different inter-channel delays. These weightings ensure that the total radiation is correct both up- and downstream, irrespectively of the choice of the inter-channel delay.

3. Adaptive design of a unidirectional two-element source

Figure 5 shows the adaptive structure proposed for the automated design of the unidirectional two-element source. The system consists of two loudspeakers (Act 1 and Act 2 in Figure 5), two microphones (Mic 1 and Mic 2), and a computer which handles the signal processing operations. A noise generator (NOISE in Figure 5) is used for producing the reference noise signal used for adaptation of the system. The adaptive algorithm is a multichannel filtered-x LMS (M-LMS) algorithm with one reference, two error, and two output signals (that is, a $1\times2\times2$ system using Kuo and Morgan's terminology [16]). Error signal $e_1(n)$ is obtained by inverting the output signal $y_{m1}(n)$ of microphone 1, but $e_2(n)$ needs to be computed as a difference of $y_{m2}(n)$ and the delayed reference signal. The M-LMS algorithm will adapt the coefficients of two FIR filters, $H_1(z)$ and $H_2(z)$, which obtain the reference signal as their input signal in the design stage.

The delay-line length Δ must be estimated in advance. This delay is needed to make the adaptive system causal: the output signal of the adaptive system must be allowed time to propagate through the duct to microphone 2 before $y_{m2}(n)$ and the reference signal are compared. The optimal estimate would be equal to the acoustic propagation delay from the source to microphone 2 plus half of the length of the adaptive FIR filters $H_1(z)$ and $H_2(z)$, when it is assumed that the filters converge to a linear-phase solution. The accuracy of the estimation of Δ is not extremely critical, however. It must only be large enough for causality. If too large a value is used for Δ, the adaptive system compensates for the extra delay by incorporating it in filters $H_1(z)$ and $H_2(z)$.

The calibration of the whole ANC system is executed in 3 phases:

1) calibration of the transfer functions $S_{11}(z)$, $S_{12}(z)$, $S_{21}(z)$, and $S_{22}(z)$ from all actuators to all microphones (see Figure 5),

Table 1. Weighting functions a and b for three different JMC-based solutions [7].

Delay optimized	a	b
downstream	$\frac{\cos(kd)}{\cos^2(kd/2)}\frac{kd/2}{\sin(kd/2)}$	$\frac{1}{\cos^2(kd/2)}$
upstream	$\frac{1}{\cos^2(kd/2)}\frac{kd/2}{\sin(kd/2)}$	$\frac{1}{\cos^2(kd/2)}$
no delay	$\frac{kd/2}{\sin(kd/2)}$	$\frac{1}{\cos(kd/2)}$

2) calibration of unidirectionality, and
3) calibration of the error path from the unidirectional source to the error detector.

Note that Figure 5 contains only the design stage of the unidirectional system, that is, phase 2.

After these procedures, the ANC operation may start. A single-channel adaptive system may be used for generating the anti-noise. The main adaptive filter that generates the anti-noise by processing the reference signal may be a separate filter as used in single-channel ANC system. Alternatively, it can be combined with the two filters of the unidirectional control system.

The major advantage obtained with the proposed adaptive design is the minimized acoustic feedback from the secondary source to the reference microphone (Mic 1). It has been shown in many former studies that the suppression of acoustic feedback is a fundamental issue in improving the performance of ANC system [3], [9].

3.1 Adaptive Swinbanks configuration

The adaptive signal processing structure shown in Figure 5 is based on the configuration of Swinbanks' method, and thus the adaptation will cause the system to automatically approach the structure of Figure 1. In addition, transfer functions $H_1(z)$ and $H_2(z)$ will contain the equalization of loudspeakers 1 and 2, respectively, in the Swinbanks' configuration. If Swinbanks' configuration is used, the common input signal for $H_1(z)$ and $H_2(z)$ may have an additional fixed integrator, according to Figure 1; Otherwise, the adaptive filters will also bring about the integration.

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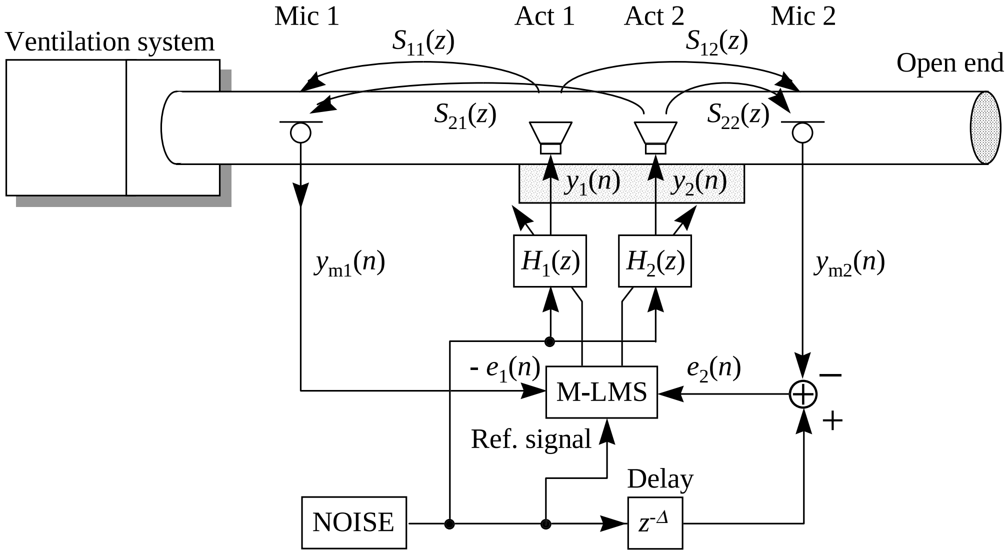

Figure 5. The adaptive signal processing structure for the automatic design a unidirectional two-element source based on Swinbanks' method. The main application for this system is ANC in a ventilation duct, as sketched in the figure.

3.2 Adaptive JMC structures

If the JMC-based approach is used, the shaded part in the middle in Figure 5 has to be substituted by the structure of Figure 6, Figure 7 or Figure 8. The structures of those figures are based on the inter-channel delay optimization downstream according to Figure 2, upstream according to Figure 3, or no inter-channel delay according to Figure 4, respectively. So the adaptation will cause the system of Figure 5 with the shaded part substituted by the structure of Figure 6, Figure 7 or Figure 8 to approach the structures of Figure 2, Figure 3 or Figure 4, respectively.

The main disadvantage of the JMC solution as compared to the Swinbanks' one is the need of separate equalization, due to using both signals (from $H_1(z)$ and $H_2(z)$) to both loudspeakers. Transfer functions $H_1'(z)$ and $H_2'(z)$ in the JMC-based configurations are used for the equalization of the frequency responses of the loudspeakers 1 and 2, respectively. They can be adjusted independently of the actual adaptive system of Figure 5 by using a separate adaptation process, but its design is beyond the scope of this paper.

The actual adaptation process of Figure 5 should cause the transfer function $H_\tau(z)$ in Figure 6 or Figure 7 to converge to the delay τ of Figure 2 or Figure 3, respectively. In the JMC-based configurations, transfer function $H_2(z)$ in Figure 5 may include a fixed integrator, according to Figure 2, Figure 3, or Figure 4.

In the case of inter-channel delay optimization down- or upstream, the JMC configurations need three transfer functions to be adapted by the M-LMS algorithms, namely $H_1(z)$, $H_2(z)$, and $H_\tau(z)$, see Figure 6 and Figure 7. If no inter-channel delay is used, only two transfer functions, $H_1(z)$ and $H_2(z)$, have to be adapted, see Figure 8. Thus the signal processing structure is simpler for the delayless case. That is also the case with off-line design [7].

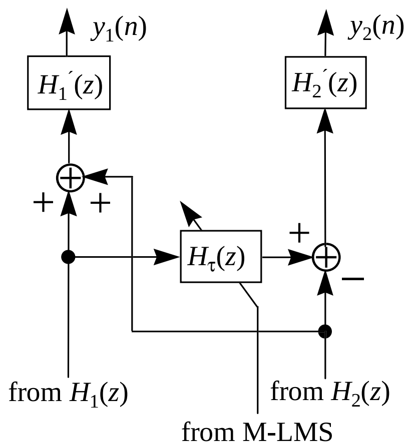

Figure 6. The shaded part of Figure 5, when the JMC-based configuration with inter-channel delay optimization downstream is used.

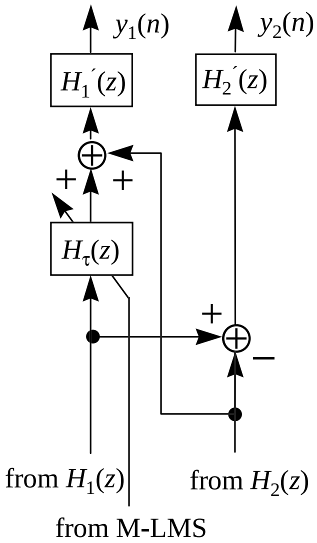

Figure 7. The shaded part of Figure 5, when the JMC-based configuration with inter-channel delay optimization upstream is used.

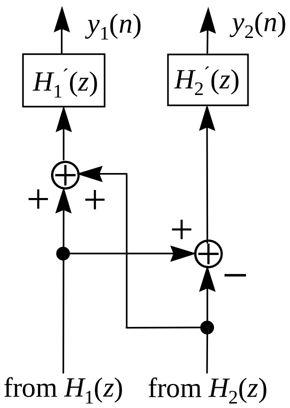

Figure 8. The shaded part of Figure 5, when the JMC-based configuration with no inter-channel delay is used.

4. Simulations

In the following we present two simulated examples of automatic design of the two-element Swinbanks source using the adaptive system proposed in Section 3. The actuators and microphones are assumed to be ideal, that is, their frequency responses are equal to unity. Also, the duct is assumed to be lossless and anechoic: the sound wave is only delayed as it travels the duct, and reflections do not occur.

The length of FIR filters $H_1(z)$ and $H_2(z)$ is 20. The adaptation constant is 0.0050 and the reference white noise signal is a sequence of 100,000 random numbers uniformly distributed in the range (–1, 1).

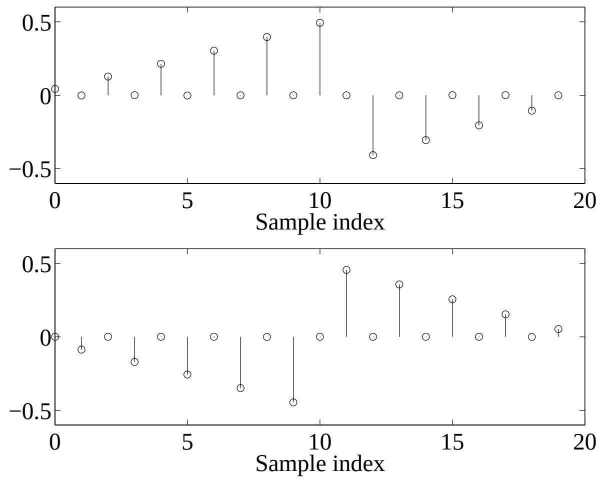

Figure 9. The impulse responses of filters $H_1(z)$ (upper) and $H_2(z)$ (lower) in example 1.

4.1 Example 1

In the first example, we have chosen the distance between the actuators to be the same as the sampling interval of the system, i.e., one unit delay. The delay between Act 2 and Mic 2 is 2 samples, and modeling delay Δ has been chosen to be 12 samples.

Figure 9 shows the impulse responses of the filters at the end of the adaptation process. These filters effectively implement the control system of the two-element Swinbanks source, that is, the integration, weighting function A, and a phase inversion between the channels are included. In addition, there is a delay between the channels that should correspond to the distance between the actuators, which in this example is 1 sample. The delay of one sample seems to have been modeled correctly (see Figure 9).

In this example, the relation between the two impulse responses is simple and it seems questionable whether 2 filters are really needed: one of the filters could easily be replaced by inverting and delaying the output signal of the other filter. However, this is caused by the simplified circumstances of the simulation. In reality, the relation of the filters is nontrivial, since the delay between the loudspeakers is not an integral multiple of the sampling interval (see [7] for a discussion).

It is of interest to consider the magnitude response of the unidirectional system in both directions. These results are shown in Figure 10. The magnitude response downstream (upper part of Figure 10) shows a considerable amount of attenuation at middle frequencies. However, at very low and

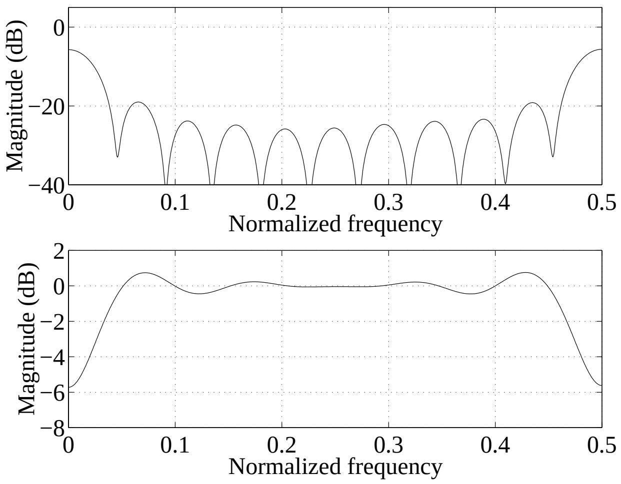

Figure 10. The magnitude frequency responses upstream (upper) and downstream (lower) in example 1.

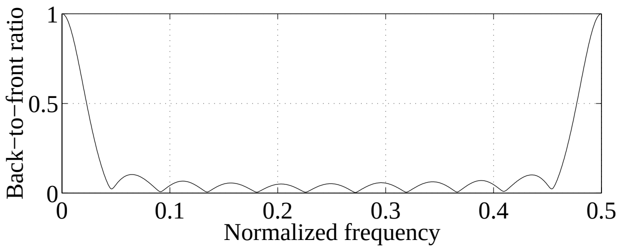

Figure 11. The back-to-front ratio in example 1.

high frequencies (with respect to the sampling rate), the attenuation is only about 6 dB. Ideally, the attenuation should be ∞ dB at all frequencies, but this is impossible due to singularities in the control functions [7]. The two filters of length 20 are able to force the response to zero at 10 frequency points only. The lower part of Figure 10 reveals that the upstream cancellation capability of the simulated system is excellent at middle frequencies, since the response is almost exactly 0 dB, which is the desired result. At very low and high frequencies, the system attenuates the signal about –6 dB.

In addition, we may consider the back-to-front ratio that has been defined by La Fontaine and Shepherd [9]. It is the ratio of the sensitivity of the system in the upstream and the downstream direction. If back-to-front ratio is 0, the system is ideally unidirectional, but if it is 1, the system is omnidirectional. In practice, we compute the back-to-front ratio by dividing the magnitude response in the upstream direction by that in the downstream direction. This result, which is shown in Figure 11, confirms that the unidirectional source works well at middle frequencies, but at very low and high frequencies, directivity has not been obtained.

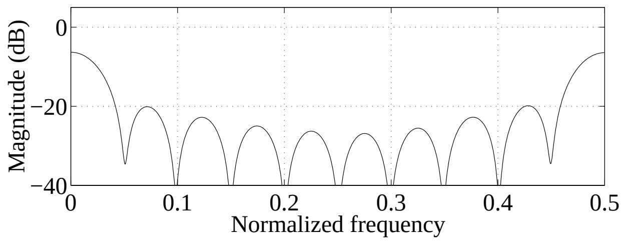

Figure 12. The obtainable sound radiation downstream when the input signal of the unidirectional system is a hypothetical ideal anti-noise signal in example 1.

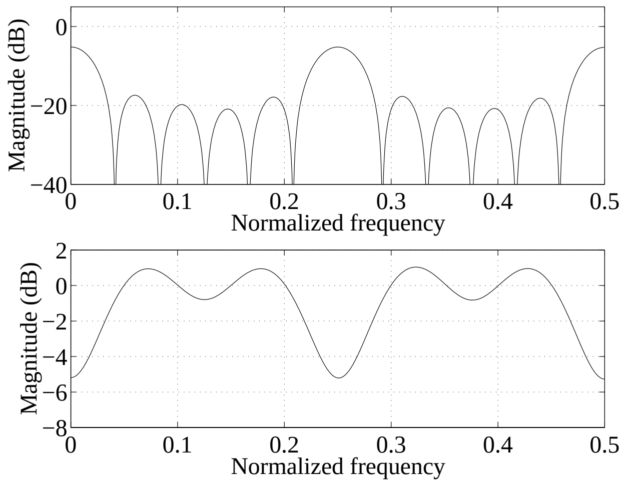

Figure 13. The magnitude responses upstream (upper) and downstream (lower) in example 2.

Figure 12 shows the downstream radiation when the direct sound is included and it is assumed that input signal of the unidirectional source is a perfect anti-noise signal. It is seen that the obtained attenuation is about the same as in the upstream direction.

4.2 Example 2

In the second example, the distance between the loudspeakers is chosen to be 2 sampling intervals. All the other parameters in the simulation are the same as in example 1. The magnitude responses upstream and downstream are displayed in Figure 13. It is seen that now it is impossible for the two-element source to achieve directivity also around the normalized frequency 0.25, since at this frequency the wavelength of sound is the same as the distance between the loudspeakers. Nevertheless, the system can achieve good attenuation upstream and desired radiation downstream at two wide frequency bands.

5. Conclusions and future work

This paper has introduced an adaptive system for the automatic design of a unidirectional source. This method is applicable to active systems for the attenuation of noise in ducts. Examples of the adaptive design of Swinbanks' source were given. Further research is needed on the adaptive design of the JMC-based variations of the unidirectional two-element source.

It is known that an ANC system affects the acoustic properties of a duct and therefore it is necessary to correct the estimates of the transfer functions $S_{ij}(z)$ as well as the error path transfer function during the cancellation process. To account for this, two approaches are available: alternation and simultaneous adaptation. By alternation we mean that the system stops the adaptation of the main adaptive filter at regular intervals and returns to adapt the other transfer function estimates. Thereafter, the adaptation of the main filter continues. Simultaneous adaptation refers to a more complicated adaptive system which updates all the transfer function estimates at the same time. The development and testing of such systems is left for future.

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