DOI: 10.61782/fa.2023.0462

STUDY OF AN ACTIVE MUTE FOR CONTROLLING THE DIRECTIVITY OF A TROMBONE

Bruno Gazengel[1*] **Christophe Ayrault**[1] **Manuel Melon**[1]
Colas Cavailles[1]

[1]Laboratoire d'Acoustique de l'Université du Mans (LAUM), UMR 6613, Institut d'Acoustique-Graduate School (IA-GS), CNRS, Le Mans Université, Avenue O. Messiaen, CEDEX 09, F-72085 Le Mans, France

ABSTRACT

In the brass instrument family, the radiated sound can be modified or attenuated using a mute, which is usually inserted in the bell of the instrument. However, this mute cannot control the directivity of the instrument. It could be interesting to develop an active mute which focuses the sound on the instrument player. The objective of this paper is to study the principle and the technological feasibility of a system of loudspeakers placed in front and around the instrument bell in order to control the directivity of the radiated sound without modifying the playability of the instrument. Assuming that the instrument can be modeled as a monopole source, simulation results show that the directivity can be controlled in the low frequency range using different layers of control sources.

Keywords: *trombone, directivity, active control, impedance*

1. INTRODUCTION

In the brass family, the trombone is a powerful instrument that can emit up to 110 dB SPL at 1 m in the fortissimo range. In order to minimize the acoustic power, different mutes are used by musicians, but they are known to alter the interaction between the instrument and the musician, making it more difficult to play certain notes. Some researchers have proposed either modifying an existing mute by adding an active control system inside the mute [1], or adding speakers near the bell to reduce acoustic power [2]. In the latter case, the proposed system consists of 8 loudspeakers placed around the bell of the trombone and 1 loudspeaker placed on axis in front of the bell. This system can reduce the radiated power in the low frequency range and has a low impact on the input impedance of the instrument. By using this system (with speakers near the bell) it is a priori possible to control the directivity of the instrument as well. To the best of our knowledge, the control of the directivity of horns has not yet been studied and could be implemented using the concepts developed so far for audio sources, e.g. the cardoid speaker. The aim of this paper is to study the feasibility of controlling the directivity of a trombone assuming that the volume velocity flowing at the bell is perfectly known. This work is a preliminary step and aims at estimating the potential of such a control.

****Corresponding author**: Bruno.Gazengel@univ-lemans.fr.*

2. SYSTEM UNDER STUDY

The system under study is composed of the trombone, called primary source, N_s control loudspeakers, called secondary sources and M observation points. The primary source is assumed to behave as a monopole source and produces a volume velocity q_p (equivalent source of the instrument). The N_s control loudspeakers (secondary sources) are assumed to be small compared to the wavelength, hence they are also modelled as point sources. They are placed close to the bell of the instrument and produce volume velocities $\boldsymbol{q_s}$. The observation points are placed at a position defined by radius $\boldsymbol{r_s}$, azimuth $\boldsymbol{\theta}$ and elevation $\boldsymbol{\phi}$. They enable to measure the pressure $\boldsymbol{p_s}$ radiated by the secondary sources and the pressure $\boldsymbol{p_p}$ radiated by the trombone. The geometry of the system is defined as shown in figure 1 (in the following, 6 secondary sources are used).

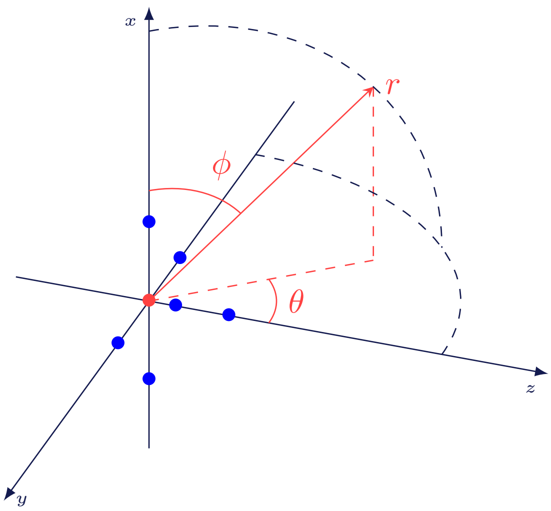


Figure 1: The studied system is composed of a primary source (●) located at the coordinate system center, 6 secondary sources (●): 4 are placed on a circle of radius $h_r = 0.16$ m in the xy plane and 2 others placed on the z-axis at $z_f = [0.05, 0.15]$ m.

3. DIRECTIVITY CONTROL

This section presents how the optimal volume flows $\boldsymbol{q_s^{opt}}$ are computed in order to achieve a target directivity for the complete system (primary and secondary sources). The pressure field radiated by all the sources at the M receivers is written:

$$\boldsymbol{p} = \begin{cases} \boldsymbol{p_p} \\ \boldsymbol{p_s} = \boldsymbol{G_s q_s} \end{cases}, \tag{1}$$

where $\boldsymbol{p_p}$ and $\boldsymbol{p_s}$ are respectively the pressure produced by the primary and the secondary sources, $\boldsymbol{G_s}$ the transfer function matrix between the secondary sources. In the following, the transfer function between the secondary source i and the receiver j is written $G_s(i,j) = j\omega\rho\frac{e^{-jkr_{ij}}}{4\pi r_{ij}}$, where ω is the angular frequency, ρ is the air density, $k = \frac{\omega}{c}$ the wavenumber with c the sound speed and r_{ij} the distance between source i and receiver j

The objective is to minimize a cost function with respect to a target pressure, defined as the pressure emitted by the primary source weighted by a target directivity function $\boldsymbol{D}$. Then, the target pressure $\boldsymbol{p_t}$ is given by:

$$\boldsymbol{p_t} = \boldsymbol{D} \circ \boldsymbol{p_p}, \tag{2}$$

where $\circ$ is the Hadamard product. Directivity $\boldsymbol{D}$ can be either defined by a continuous function of space, for example $\boldsymbol{D} = \frac{1+\cos\boldsymbol{\theta}}{2}$, which corresponds to a cardioid patten, or by a discontinuous function of space, called gate function, for which $\boldsymbol{D} = 1$, $\forall\theta_{min} < \theta < \theta_{max}$ and $\forall\phi_{min} < \phi < \phi_{max}$ and $\boldsymbol{D} = 0$ elsewhere. In the case of gate functions, the secondary sources are constrained not to radiate in a zone (called bright zone where $\boldsymbol{p_p}$ is emitted) and produce the opposite of the primary source pressure in another zone (called dark zone), as depicted in figure 2. This approach is similar to the one used for creating personal sound zones [3], especially the forced pressure matching algorithm [4].

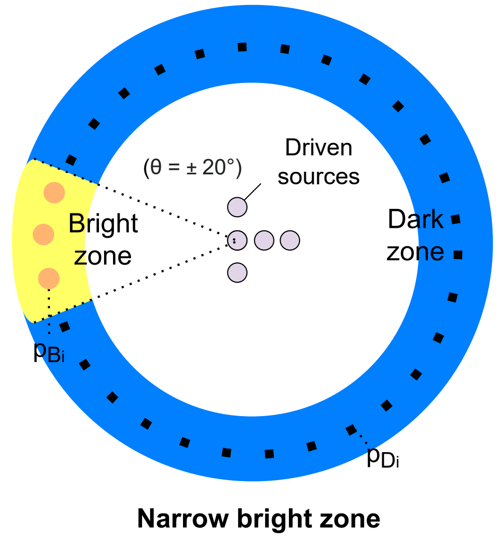


Figure 2: 2D plot of the target directivity defined by a gate function: the bright zone is colored in yellow and the dark zone in blue. Here, the bright zone is smaller than the dark zone ($\theta_{min} = 160°$, $\theta_{max} = 200°$).

The target pressure $\boldsymbol{p_{ts}}$ for the secondary sources can be written as:

$$\boldsymbol{p_{ts}} = \boldsymbol{p_p D} - \boldsymbol{p_p} = \boldsymbol{p_p}(\boldsymbol{D} - 1). \tag{3}$$

The cost function to minimize is thus defined by:

$$J = (\boldsymbol{p_{ts}} - \boldsymbol{G_s q_s})^H(\boldsymbol{p_{ts}} - \boldsymbol{G_s q_s}) + \beta\boldsymbol{q_s}^H\boldsymbol{q_s}, \tag{4}$$

with a regularization term β added to limit the loudspeaker effort and therefore reduce the matrix conditioning [5].

The optimal secondary volume flow $\boldsymbol{q_s^{opt}}$ is obtained by differentiating the quadratic function of equation 4 with respect to $\boldsymbol{q_s}$ [2, 6] and generates a unique solution given by:

$$\boldsymbol{q_s^{opt}} = \left(\boldsymbol{G_s}^H \boldsymbol{G_s} + \beta \boldsymbol{I}\right)^{-1} \boldsymbol{G_s}^H \boldsymbol{p_{ts}}. \tag{5}$$

Two indicators are defined to evaluate the performance of the control according to the needs of the targeted application.

The first one is the weight of the filter $\boldsymbol{w} = \frac{\boldsymbol{q_s^{opt}}}{q_p}$ informing about the volume velocity that needs to be produced by each control source (in practice by each secondary loudspeaker).

The second indicator is the error between the pressure resulting from the control and the target pressure. The normalized average spatial error of the control between the desired and the reproduced sound field is expressed as [7]:

$$\overline{e} = \frac{(\boldsymbol{p_t} - \boldsymbol{p})^H (\boldsymbol{p_t} - \boldsymbol{p})}{\boldsymbol{p_t}^H \boldsymbol{p_t}}. \tag{6}$$

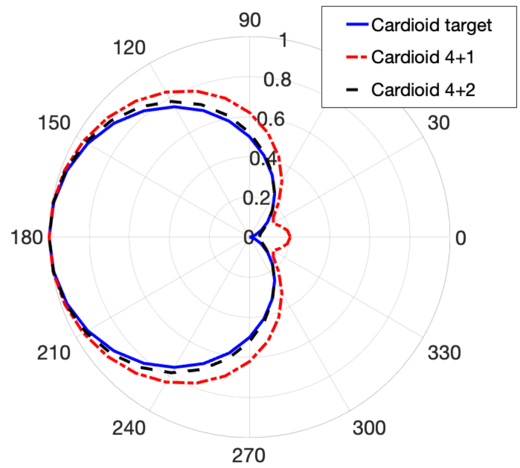
The smaller the error $\overline{e}$, the more the pressure field resulting from the control matches the target pressure.

4. EXPECTED PERFORMANCES

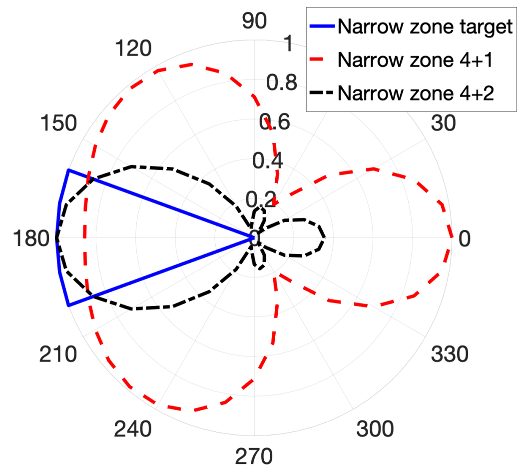
This part deals with the impact of the target directivity $\boldsymbol{D}$ on the control efficiency. For this study, 4 secondary sources are placed at ($z_f = 0$, $h_r = 0.16$) m and 1 or 2 secondary sources ($z_f = 0.05, 0.15$ m) are placed in line on the bell axis as shown in figure 1. $M = 614$ observation points ($\Delta\theta = \Delta\phi = 10°$) are used to estimate the optimal volume velocity $\boldsymbol{q_s^{opt}}$. This configuration is inspired from a previous work dealing with the control of acoustic power [2] in which different secondary sources are placed around the trombone bell and one is placed on the bell axis.

Two target directivities are studied as shown in figure 3 (blue curves), the bell of the trombone being directed towards 0°. The first is a cardioid $\boldsymbol{D} = \frac{1+\cos\boldsymbol{\theta}}{2}$, the second is a narrow bright zone defined by $\boldsymbol{D} = 1$ for $\theta \in [160°, 200°]$ and $\phi \in [70°, 110°]$. All the simulations are performed using $\beta = 0$ (equation 5). A constant regularization term [2, 8, 9] could be used in experiments so that the nominal power of the control loudspeakers and their maximum excursion are not exceeded.

Figures 3 and 4 show the target directivities and the directivities obtained after control respectively at 200 Hz and 1000 Hz. The target cardioid directivity is reached at 200 Hz but not at 1 kHz, the number of control sources being too small. Trying to get a narrow bright zone is difficult for the system having a small number of control sources. Figure 5 shows the error $\overline{e}$. It shows that narrow bright zone target directivity leads to a larger error even when using 2 sources instead of 1 on the z axis.



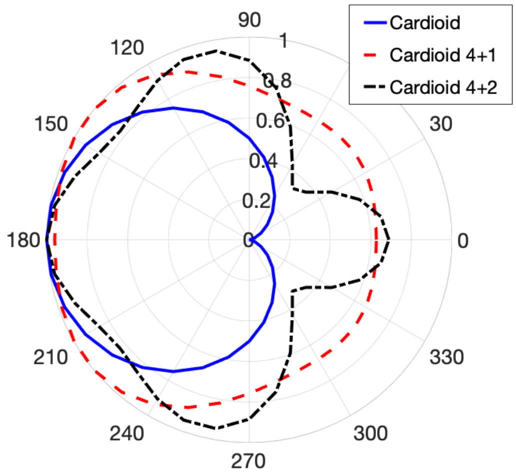
(a) Cardioid target directivity.



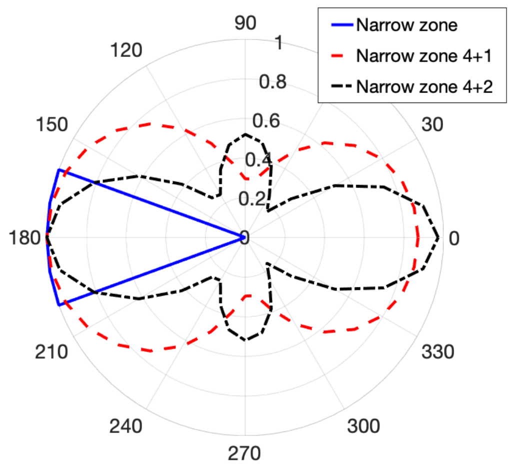
(b) Narrow bright zone target directivity.

Figure 3: Target directivities and directivities obtained with optimisation for 2 configurations (4 sources around the bell + 1 or 2 sources in front of the bell hereafter referred as 4+1 or 4+2) at 200 Hz.

It is shown in Ref. [2] that for acoustic power control,



(a) Cardioid target.



(b) Narrow bright zone target.

Figure 4: Target directivity and directivity obtained with optimisation for 2 configurations (4 sources around the bell + 1 or 2 sources in front of the bell hereafter referred as 4+1 or 4+2) at 1000 Hz.

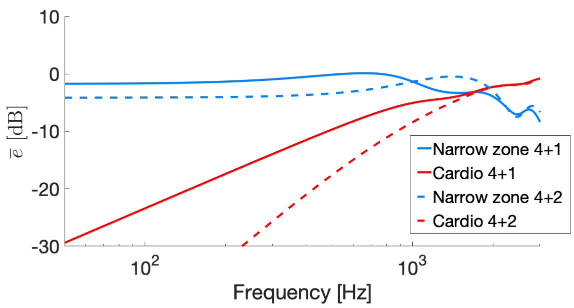


Figure 5: Normalized average spatial error $\overline{e}$ between the target directivity and the pressure resulting from the control when the bright zone is narrow, when the target is a cardioid.

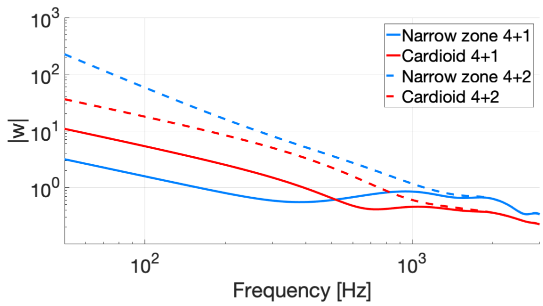


Figure 6: Control filter modulus of the secondary source closest to the primary source for a control imposing a narrow bright zone and a cardioid.

the secondary source closest to the primary source ($z_f = 0.05$ m) is the one that requires the largest control filter magnitude $|w|$ to perform the control. For this reason, the modulus of the control filter to be applied to the closest secondary source is observed in figure 6.

These results show that the amplitude of filter $|w|$ is large in low frequency for every configuration, so that in the following the analysis is done at a frequency of 50 Hz.

This target directivity requires a low amplitude for the first source on the z axis when using $4 + 1$ control sources ($|w| \simeq 3$) but needs a very large amplitude in the low frequency range for $4 + 2$ secondary sources ($|w| \simeq 200$), which means that this control would be very difficult to realize and would lead to a low efficiency.

When the target directivity is a cardioid, the error is low when using 4 sources around the bell and 1 on the z axis (4 +1), the filter amplitude being $|w| \simeq 10$ and the control is efficient up to around 1 kHz. Adding a source on the z axis (4 + 2 sources) lowers the error but needs more effort (filter amplitude higher with $|w| \simeq 35$) and enables to enlarge the bandwidth up to 1.5 kHz.

Figure 7 shows the filters weight w for configuration 4 + 2 sources when a cardioid is used as a target directivity. Simulation results show that the secondary source placed at $z_f = 0.15$ m has also a high control

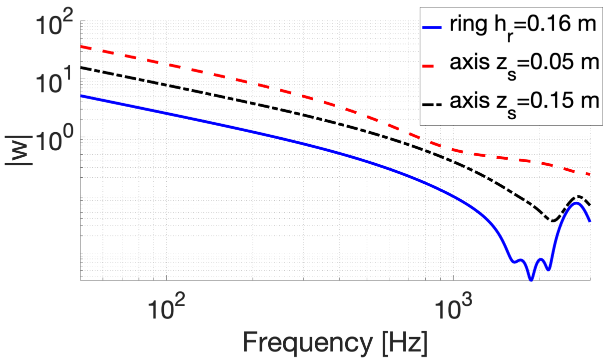


Figure 7: Control filter modulus of the secondary sources placed on ring ($z_f = 0,\ h_r = 0.16$ m) and on the z axis (source 1 at $z_f = 0.05$ m, source 2 at $z_f = 0.15$ m) imposing a cardioid directivity for configuration 4 + 2 control sources.

filter modulus but needs to produce a volume velocity which is 2 times less important than the volume velocity of source 1 in low frequency. A less important effort is imposed on the sources placed around the bell ($h_r = 0.16$ m, $z_f = 0$) which have a reduced importance for the control (7 times less important than the volume velocity of source 1) in low frequency (50 Hz).

Finally, when the directivity is of a lower order of complexity, the control is easier to achieve. For the cardioid target directivity, the error is minimized using 2 sources on the z axis but this enhances the amplitude of the source placed near the bell, suggesting using a powerfull source or many physical sources instead of one near the bell. The other sources (around the bell, second source on the z axis) do not need to produce a high volume velocity.

5. TOWARDS A REAL APPLICATION

In a real application case (trombone), the voltage applied to control loudspeakers depends on the trombone volume velocity spectrum with $q_p \simeq 1$ litre/s at 50 Hz, $q_p \simeq 5$ litre/s at 500 Hz [2] and on the loudspeaker characteristics. The main constraints concerns the control source 1 ($z_f = 0.05$ m) at 50 Hz. Assuming that this source is an ideal loudspeaker having the same diameter of the trombone bell (20 cm), the membrane peak displacement needs to be 1 mm using 4 +1 control sources and 3.8 mm using 4 + 2 control sources. This technical solution seems to be feasible but a large membrane would modify strongly the bell radiation. For a smaller membrane, the excursion will be much increased and should be limited using a regularisation term (β in equation 5).

6. CONCLUSION

A directivity control of the trombone radiation is achievable considering a target directiviy with a smooth spatial variation (cardioid pattern for example). This can be done by placing several layers of control sources depending on the required power and control frequency band. The simulation results show that two or three layers of control sources should be used. First a control source generating a very high volume velocity has to be placed in front of the bell. Second, another source can be placed on the bell axis in order to minimize the spatial error. Finally in this work a ring of sources generating a low volume velocity is placed around the bell and contributes to the optimization with non zero filters.

For future work the effect of control sources number on the error and effort should be studied more in detail in order to define an optimal configuration defined as a compromise between error and effort. The effect of the control on the instrument input impedance should also be investigated.

7. ACKNOWLEDGEMENTS

The authors thanks Joël Gilbert and Emmanuel Brasseur for measurements performed on a real trombone (figure 8).

8. REFERENCES

[1] T. Meurisse, A. Mamou-Mani, R. Caussé, B. Sluchin, and D. B. Sharp, "An active mute for the trombone," *The Journal of the Acoustical Society of America*, vol. 138, pp. 3539–3548, Dec. 2015.

[2] C. Cavaillès, B. Gazengel, M. Melon, and C. Ayrault, "Study of the optimal active control of a trombone," *Acta Acustica*, vol. 6, p. 18, 2022.

[3] T. Betlehem, W. Zhang, M. A. Poletti, and T. D. Abhayapala, "Personal Sound Zones: Delivering interface-free audio to multiple listeners," *IEEE Signal Processing Magazine*, vol. 32, pp. 81–91, Mar. 2015.

Figure 8: Experimental estimation of the trombone volume velocity.

[4] L. Vindrola, M. Melon, J.-C. Chamard, and B. Gazengel, "Pressure matching with forced filters for personal sound zones application," *Journal of the Audio Engineering Society*, vol. 68, no. 11, pp. 832–842, 2020.

[5] P. A. Nelson, "Estimation of Acoustic Source Strength by Inverse Methods: Part I: Conditioning of the Inverse Problem," tech. rep., 2000.

[6] P. A. Nelson, A. Curtis, S. Elliott, and A. Bullmore, "The minimum power output of free field point sources and the active control of sound," *Journal of Sound and Vibration*, vol. 116, pp. 397–414, Aug. 1987.

[7] J.-W. Choi, Y. Kim, S. Ko, and J.-H. Kim, "Super-Directive Loudspeaker Array for the Generation of Personal Sound Zone," in *The Journal of the Audio Engineering Society*, (San Fransisco, USA), Audio Engineering Society, Oct. 2008.

[8] P. A. Nelson and S. Yoon, "Estimation of acoustic source strength by inverse methods: part I, conditionning of the inverse problem," *Journal of Sound and Vibration*, vol. 233, pp. 643–668, Dec. 1999.

[9] S. Yoon and P. Nelson, "Estimation of acoustic source strength by inverse methods: part II, experimental investigation of methods for choosing regularization parameters," *Journal of Sound and Vibration*, vol. 233, pp. 669–705, Dec. 1999.