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**Physics of Musical Instruments and the Voice:
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**Improving the stability of a hybrid wind instrument using
two microphones**

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Abstract

A hybrid wind instrument is constructed by putting a theoretical excitation model (such as a real-time computed physical model of a clarinet embouchure) in interaction with a real wind instrument resonator.

In previous work, the successful construction of a hybrid wind instrument has been demonstrated, with the interaction facilitated by a loudspeaker and a single microphone placed at the entrance of a clarinet-like tube. The prototype was evaluated using physical models of a single-reed, a lip-reed and a bow-string interaction. Musically relevant results were obtained when the negative gradient of the nonlinear excitation function was limited to a certain threshold. When surpassed, erroneous noises appeared.

In the present paper, a study of the open-loop system (the input-to-output response excluding the excitation model) reveals that this instability is caused by strong, high-frequency resonance peaks combined with an inverted phase response. The high frequency resonance peaks appear to result from non-planar air vibration modes in the small cavity in front of the loudspeaker. Hence, they are avoided by repositioning the microphone at the centre of the loudspeaker cavity. Meanwhile, the inverted phase state occurs due to various phase lag sources such as the inevitable input-to-output latency of the computing system. This is accounted for by introducing a second microphone a distance $c \cdot \Delta t$ along the tube (where c is the speed of sound and Δt the latency).

The excitation models are implemented on a new digital real-time audio platform, "Bela", supporting multiple audio inputs. A better stability is obtained and evaluation with a real clarinet gives musically relevant results.

Keywords: hybrid, wind instrument, two-microphone, stability

Improving the stability of a hybrid wind instrument using two microphones

1 Introduction

The development and evaluation of a hybrid wind instrument using a loudspeaker has been described in earlier work [1, 2]. Figure 1 explains the concept: a physical “excitation model” (for instance a single-reed embouchure) is simulated on a computer and interacts with a real acoustical resonator (a clarinet-like tube) so that the whole is able to generate hybrid self-sustained sounds. A microphone picks up the pressure at the entrance of the tube and sends the signal to the computer. Using the physical model of a single-reed embouchure, the corresponding flow rate signal is calculated and (after some processing) sent to a loudspeaker, which will produce a real acoustic flow at the tube entrance. This feedback system functions similarly to a normal clarinet, but in a hybrid manner. It should be noted that for a coherent functioning, a “real-time” computer is required, with an input-to-output latency that is much smaller than the period of the oscillations that the hybrid instrument will produce.

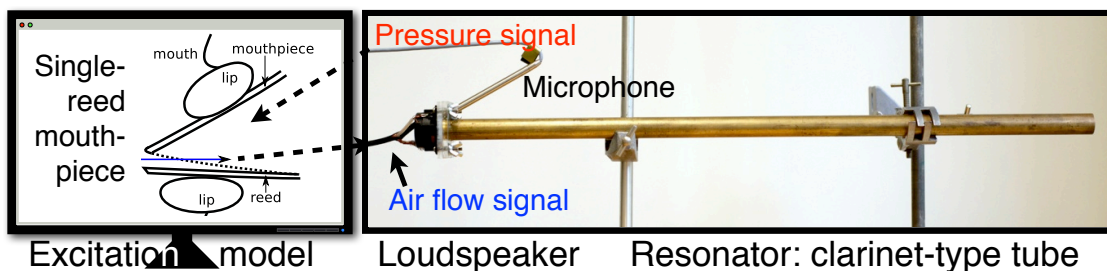


Figure 1: The hybrid wind instrument set-up: a computed excitation model (here a single-reed embouchure model) in interaction with a physical resonator by means of a loudspeaker and a microphone.

A hybrid instrument supports two main research interests. First, the device’s potential as a musical instrument can be explored, mostly from the timbre perspective, which is an active musical focus of today. Here, the same control precision can play a role in the accessibility of certain (variations of) sounds. While the computed environment allows the modelling of any conceivable excitation and also handles electronic parameter variations, the physical control over the resonator (the fingering) remains, which opens up an alternative range of musical expression with the advantage of relatively low computational power needs.

Second, imagining this instrument in the context of acoustic wind instrument research, it would be of substantial value to have a repeatable and precisely quantified control over an exciter that is linked to a resonator of interest. This matches with the objectives of the now well-established “artificial mouths” for wind instruments (e.g. [3]).

Only minor contributions on the hybrid wind instrument concept have been reported to date. Maganza first briefly explored a set-up [4] and since then a small number of works on closely

related subjects have been carried out, for example [5, 6, 7]. More recently, an identical approach has been implemented, but using an electrovalve as flow actuator [8]. For our hybrid instrument, a loudspeaker is used to perform the actuation and it has been shown that, by introducing some correcting filters, a very good accuracy can be achieved, so that realistic clarinet tones can be produced [1, 9].

In the present paper, the initial prototype is described in section 2, and it is explained how unstable behaviour can appear. Then, in section 3, the development and evaluation of an improved prototype is laid out.

For the sake of simplicity and generality, dimensionless quantities are used throughout this paper. For instance, a dimensionless pressure \bar{p} is related to the pressure p , by $\bar{p} = p/p_M$, where p_M is the “beating pressure”, characteristic to the embouchure model. Similarly, flow rates are non-dimensionalised by the relation $\bar{q} = q Z_c/p_M$, where Z_c is the characteristic impedance of the resonator.

2 The initial prototype and its instabilities

2.1 Set-up using a single microphone

The design of the initial prototype is shown schematically in figure 2. The prototype features a loudspeaker and a single microphone, both positioned at the entrance of the tube. Given that the loudspeaker’s membrane velocity is transformed by the driver’s dynamics (which can be modelled as a mass-spring-damper system), a feedback and feedforward filter are implemented on the computing system to account for the loudspeaker’s response. Details of the filters are reported in an earlier paper [1].

The pressure signal captured by the microphone is sent to the computing system which introduces input-to-output (I/O) latency. For the case of the computing system used for the initial prototype, the latency is $20 \mu\text{s}$, corresponding to 1 sample ($N = 1$) at a sampling rate of $f_s = 50 \text{ kHz}$. The dimensionless pressure signal is sent to both the feedback filter and the excitation model. For the latter, the (equally delayed) dimensionless flow rate signal is subtracted, in order to obtain the required (dimensionless) “historical pressure” $\tilde{\bar{p}}_h = \bar{p}(z^{-1}) - \tilde{\bar{q}}(z^{-1})$ (the tilde sign is used for the approximate historical pressure supplied to the excitation model, and for the flow rate calculated by this model).

For a coherent hybrid operation, the flow rate generated by the loudspeaker should be as close as possible to the flow rate calculated by the excitation model and then sent to the feedforward filter, so that $\bar{q} \approx \tilde{\bar{q}}$; and the latency should be as small as possible so that $\tilde{\bar{p}}_h \approx \bar{p} - \bar{q}$.

2.2 Observed instabilities with a single-reed excitation model

2.2.1 The single reed excitation model

By introducing a single-reed excitation model to the set-up, hybrid self-sustained clarinet sounds can be produced. We adopted a classical model, initially proposed by Wilson and Beavers [3]. The detailed description of this model is not needed here but the resulting nonlinear curve,

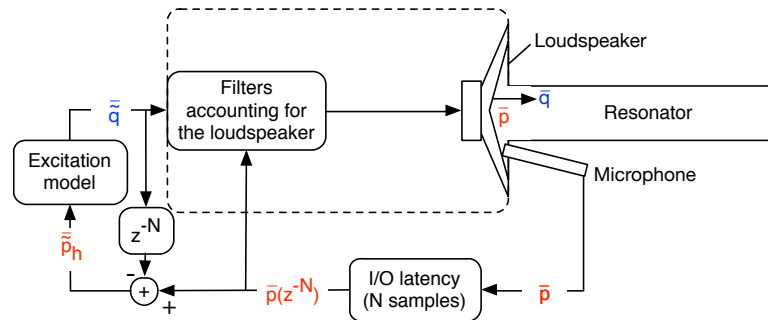


Figure 2: **Schematic diagram of the initial prototype, using a single microphone to supply a pressure signal to the excitation model. The pressure is delayed by the input-to-output latency of the computer.**

relating the outputted flow rate as a function of the pressure in the mouthpiece $\bar{q}(\bar{p})$, is represented as a dashed curve in figure 3. γ represents the dimensionless mouth pressure and ζ is a generalised “embouchure parameter”, which controls the vertical scaling of the nonlinear curve.

It can be verified for instance that, when the mouthpiece pressure is equal to the mouth pressure, i.e. $\bar{p} = \gamma$, there is no flow. Alternatively, when the mouthpiece pressure is much lower than the mouth pressure, the reed hits the lay and there is also no flow. It is between these two states that an air flow will enter the resonator, so that at each mouthpiece pressure transition to the opposite pressure value, there will be a puff of air that will contribute to the acoustic energy inside the resonator to maintain a self-sustained oscillation.

Given that the flow rate instantaneously influences the pressure at the entrance of the tube, \bar{p} is also an instantaneous function of \bar{q} . Hence, for a numerical sequential simulation, it is not possible to directly use the implicit equation $\bar{q}(\bar{p})$ to obtain the flow rate. Therefore, the so-called “historical pressure” is used, which excludes the pressure contribution from the flow rate: $\bar{p}_h = \bar{p} - \bar{q}$. Using an approach proposed by Guillemin et al. [10], an explicit equation $\bar{q}(\bar{p}_h)$ is derived, whose (quasi-statically approximated) curve is represented as a solid line in figure 3.

2.2.2 Observed instabilities

Although a stable and musically relevant self-sustained operation was obtained for a range of ζ values, when $\zeta \geq 0.35$ an unstable behaviour appeared. Figure 4 shows the (steady state) pressure wave generated using the single-reed excitation model with $\zeta = 0.35$ and $\gamma = 0.5$. As can be seen, a high frequency oscillation appears when the mouthpiece pressure is around $\bar{p} = 0.5$. Looking up this pressure value on the nonlinear curve leads to the hypothesis that the system becomes unstable when a strong negative gradient is encountered on the curve.

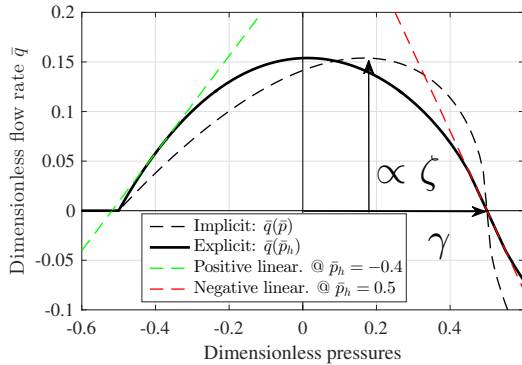


Figure 3: **Characteristic nonlinear curves associated with the functioning of the embouchure.** The dashed curve corresponds to the implicit equation $\bar{q}(\bar{p})$ and the solid curve to the explicit equation $\bar{q}(\bar{p}_h)$. γ represents the dimensionless mouth pressure and ζ is a global “embouchure parameter”.

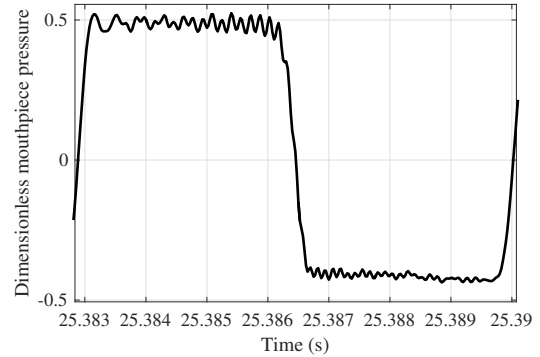


Figure 4: **Pressure wave of a self-sustained oscillation produced with the initial prototype (with applied parameters $\zeta = 0.35$ and $\gamma = 0.5$), demonstrating the appearance of a high frequency oscillation.**

2.3 Stability analysis of a linearised system: study of the open-loop

While it is not straightforward to derive theoretical stability criteria in a nonlinear system, it is possible to linearise the nonlinear curve at an arbitrary point so that a locally valid linear stability study can be carried out. Considering the nonlinear single-reed curve, it can be seen that a positive and a negative linearisation is possible, as is shown by respectively the green and red dashed lines in figure 3.

A typical stability study in such a system involves considering the Bode plot of the “open-loop”, i.e. the system contained between the signals \bar{q} and \bar{p}_h , including the filters, the coupled loudspeaker-resonator system, the microphone and the calculation of \bar{p}_h . In the frequency domain, this system is represented by the transfer function \bar{P}_h/\bar{Q} and its Bode plot (obtained by supplying a sine-swept signal \bar{q} while recording \bar{p}_h) is shown in figure 5.

The gain margins are the gains at the 0° and -180° phase transitions, respectively corresponding to the positive and negative linearisations of the excitation curve (indicated in respectively green and red in figure 5). The stability criterion stipulates that at frequencies for which the gain multiplied by the slope of the linearised excitation curve surpasses 1, the system is unstable.

While the low frequency positive instabilities at the resonator’s resonance frequencies enable the desired instability that leads to the self-sustained operation, there are also a few prominent negative instabilities at high frequency which can be noted (these are normally not present in the case of an entirely acoustic instrument whose open-loop phase response remains in the

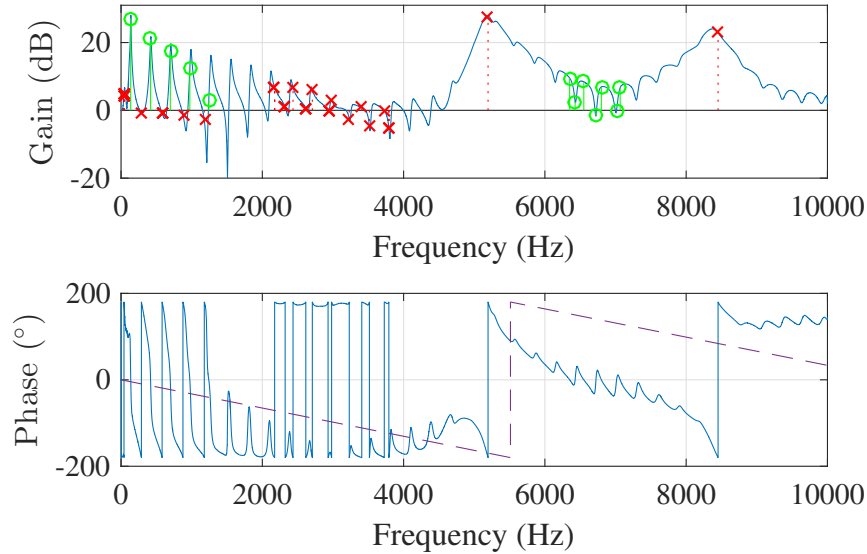


Figure 5: **Open-loop stability study: Bode plot of the ratio \bar{P}_h/\bar{Q} for the initial prototype; with indication of the 0° and -180° gain margins. The dashed line represents the phase response corresponding to the I/O latency of the computing system.**

[$90^\circ, -90^\circ$] range).

Empirical evaluations revealed that the high gains at these instabilities result from non-planar vibration modes in the small cavity in front of the loudspeaker, which has a diameter that is about 2.7 times the diameter of the tube. Meanwhile the observed increasing phase lag has multiple causes such as the I/O latency (whose phase response is shown as a dashed line in figure 5) and loudspeaker effects which could not be accounted for.

3 The improved prototype

In order to improve the stability, a new prototype with a modified set-up has been constructed. For this prototype, the digital real-time audio platform “Bela” is used (designed by Andrew McPherson et al. at Queen Mary University [11]). This computing system features several analogue inputs and has a minimum total I/O latency of $\Delta t = 90.7 \mu\text{s}$, with $N = 4$ and $f_s = 44.1 \text{ kHz}$.

3.1 Accounting for the instabilities

By introducing a second microphone a distance $c \cdot \Delta t$ along the tube (where c is the speed of sound), the I/O latency can be accounted for. (This is a similar approach to that of Guicking [12], who used two microphones to modify the reflection coefficient.) Figure 6 shows a schematic diagram explaining this concept by considering the pressure as subdivided forward and backward travelling waves, respectively represented by \bar{p}^+ and \bar{p}^- .

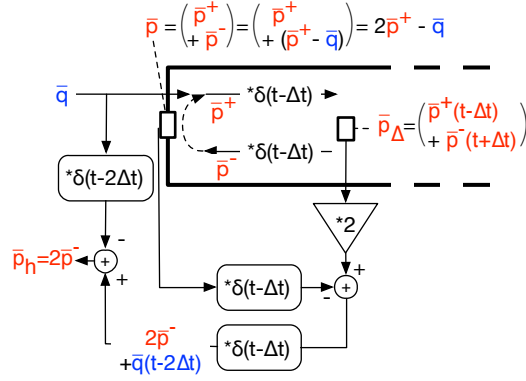


Figure 6: Schematic diagram describing how the historical pressure \bar{p}_h can be obtained from two pressures \bar{p} and \bar{p}_Δ , which are delayed by respectively $2\Delta t$ and Δt . \bar{p} is measured at the (closed) tube entrance and \bar{p}_Δ at a distance $d = c.\Delta t$ from the entrance.

Hence, from the pressure signal at the first microphone one can derive:

$$\bar{p} = \bar{p}^+ + \bar{p}^- = \bar{p}^+ + (\bar{p}^+ - \bar{q}) = 2\bar{p}^+ - \bar{q} \Rightarrow \bar{p}^+ = \frac{\bar{p} + \bar{q}}{2} \quad (1)$$

As such, the pressure at the second microphone position can be expressed as:

$$\bar{p}_\Delta = \bar{p}^+(t - \Delta t) + \bar{p}^-(t + \Delta t) = \frac{\bar{p}(t - \Delta t) + \bar{q}(t - \Delta t)}{2} + \bar{p}^-(t + \Delta t) \quad (2)$$

Finally, the historical pressure can be obtained from delayed measurements from both microphones and a delayed flow rate signal:

$$\bar{p}_h = 2\bar{p}^- = 2\bar{p}_\Delta(t - \Delta t) - \bar{p}(t - 2\Delta t) + \bar{q}(t - 2\Delta t). \quad (3)$$

This principle is then applied in the final implementation (schematically represented in figure 7). The distance of the two microphones is chosen so as to correspond to the I/O latency: $d = c.\Delta t = c.N/f_s$. A second improvement simply involves the repositioning the first microphone at the centre of the loudspeaker cavity, so that the pressure nodes of the non-planar vibration modes are avoided.

3.2 Study of the open-loop

Figure 8 shows the transfer function $\tilde{\bar{P}}_h/\tilde{\bar{Q}}$, corresponding to the new prototype. As can be seen, the high frequency gain peaks caused by the non-planar modes have disappeared and the phase lag is reduced so that, all together, the gain margins are decreased and the system remains stable, even in regions where the gradient of the nonlinear curve becomes steep.

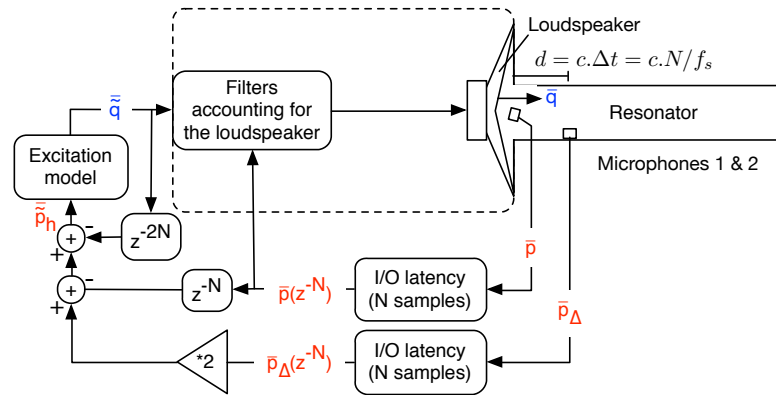


Figure 7: **Schematic outline of the implementation of the new prototype, using two microphones to obtain $\bar{p}_h \approx \bar{p}_h (= \bar{p} - \bar{q})$, which is an approximation that is free of I/O latency.**

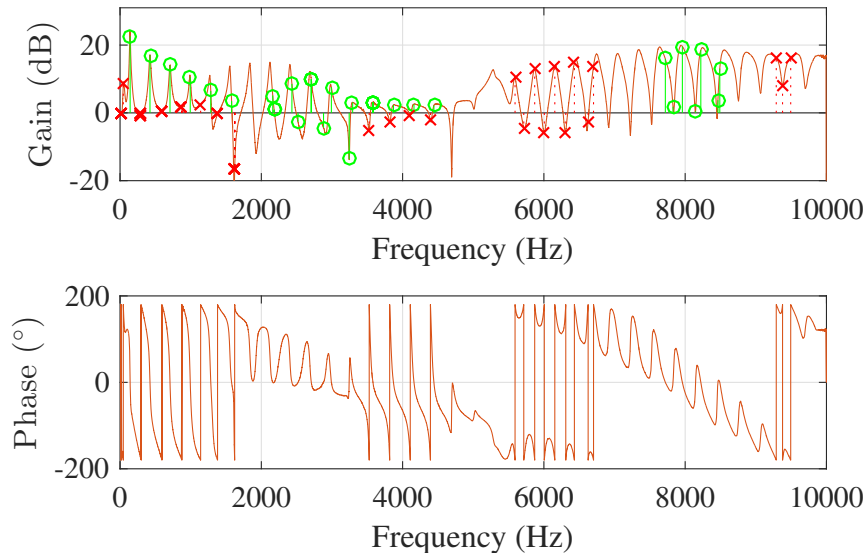


Figure 8: **Open-loop stability study: Bode plot of the ratio \bar{P}_h/\bar{Q} for the improved prototype; with indication of the 0° and -180° gain margins.**

However, it should be noted that the lower frequency response (around 2 kHz) has also changed somewhat. By studying the influence of separate components, it has been shown that this is a result of the two-microphone concept, combined with the loudspeaker front cavity, which introduces a small phase shift compared with a rigidly terminated tube.

3.3 Self-sustained operation

An evaluation of the improved prototype was carried out, using the single-reed excitation model with the same parameter values. The resulting pressure wave, represented in figure 9, is indeed free of high frequency noise. The system remains stable for ζ values up to 0.7, a wider

timbre domain can be reached and an evaluation with a real clarinet resonator results in musically relevant sounds. This opened up the possibility of studying the particular phenomenon of “period doubling bifurcations”, which is a form of multiphonic where subharmonic frequencies are introduced and which is only manifested when $\zeta \geq 0.4$. However, it is important to note that the wave is less square in nature than it was before (and as also obtained with simulations), which is most likely due to the change in the lower frequency region of the open-loop.

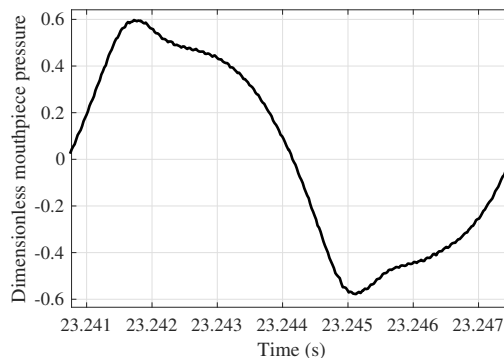


Figure 9: **Pressure wave of a self-sustained oscillation produced with the improved prototype (with a single-reed embouchure model with parameters $\zeta = 0.35$ and $\gamma = 0.5$). No high frequency noise is present, but the wave is less square in nature.**

4 Conclusions

It has been demonstrated that the behaviour of an initial prototype of a hybrid wind instrument (using a single-reed excitation model) can become unstable when the nonlinear curve of the excitation model has regions with strong negative gradients.

A linearised “open-loop” study on the \tilde{P}_h/\tilde{Q} transfer function Bode plot revealed that this was due to low gain margins. The high frequency gain peaks appeared to result from non-planar air vibration modes in the front-loudspeaker cavity, which can be avoided by repositioning the microphone at the centre of the loudspeaker cavity. Meanwhile, the inverted (-180°) phase state was due to various phase lag sources, including the input-to-output latency of the computing system. This could be accounted for by introducing a second microphone a distance $c \cdot \Delta t$ along the tube.

Those changes have been implemented in a new prototype, which uses the “Bela” real-time audio platform. While this computing system has a higher input-to-output latency, it also features several analogue inputs so that the two-microphone method can compensate for this delay. The new prototype’s open-loop study revealed improved gain margins, but at expense of a modified lower frequency response.

Evaluation with the single-reed model resulted in more stable but less coherent (i.e. less square-shaped) sounds, enabling a larger variety of musical sounds and opening up the study of the period doubling phenomenon.

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