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Attack transients in a loudspeaker / resonator coupled system

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ABSTRACT

Electroacoustic systems have been used in the past to simulate the self oscillations produced by a mechanic system like a violin string or wind instruments. For a wind instrument, the exciter block can be replaced by an electric circuit providing energy and a non-linear feedback to the acoustic resonator. These systems can be simpler to model than the real instrument, while allowing a general behaviour close to that of the original. Compared to a numerical simulation, this system allows to maintain the exact behaviour of the resonator while the shape of the non-linear function relating incoming and outgoing pressure waves can be finely adjusted, or at least known with great precision. This knowledge however requires a good choice of electroacoustic transducers and a good description of its coupling to the acoustic resonator. After a brief description of the system and its modes of resonance, this presentation will focus on the application of such a system to the study of attack transients. In particular, the interest is to predict the regime the instrument will stabilise into after a particular attack envelope imposed to its control parameters.

MOTIVATION

An electro-acoustic instrument combines mechanical vibrations to electrical currents that can be transformed through a circuit to modify the sound. In most popular cases, the electrical signal is a product of the mechanical vibration but does not interact with it. In the case of an auto-oscillating instrument, the electric component can replace part of the acoustic signal emulating or modifying it. For example, Maganza et al. (1986) worked on a system replacing the exciter by a digital circuit and Kitano et al. (1983) used a similar electroacoustic system to study bifurcations in an electro-acoustic system, that will be referred to as the *electro-acoustic clarinet*

The use of an electro-acoustic system instead of a real wind instrument is justified by the fact that the exciter system is easier to characterise mathematically, and the number of parameters that control it can be significantly reduced and simplified.

Another advantage is that the variables that characterise the system are easy to capture, allowing to follow its evolution since the start of the oscillation. This approach can be seen as a complement to time-domain simulations of the instrument [Vergez et al. (2003)], with the advantage that the complex response of the bore, in particular on fingering transitions, does not need to be modeled. In future this system could validate some of the aspects of finger coordination recently studied [Almeida et al. (2009)].

Although the motivation of the project was at start to model a clarinet system, in the following paragraphs we will show that the electro-acoustic clarinet introduces a few complications that need to be correctly modeled before building a complete electro-acoustic system.

This work will follow a more fundamental approach, describing the modes of auto-oscillation that the system naturally follows. We will then present an experiment on the transient part of the oscillation. Even if the characteristic non-linearity of a reed instrument is not introduced, a linear system can be seen as a limit of the reed instrument for low amplitudes.

INTRODUCTION AND CONTEXT

The electro-acoustic clarinet described here consists of the bore of an acoustic instrument (or any other acoustic resonator) coupled to an electroacoustic block that replaces the usual exciter (a clarinet mouthpiece for instance). This block includes two transducers (a microphone and a loudspeaker) that convert between acoustic pressure or velocity into an electric current or tension. Connecting the two transducers is a circuit that can supply energy to feed an auto-oscillation in the bore.

Although the clarinet involves some complexity in the definition of the non-linear function relating the measured and excited pressures, an auto-oscillation can be produced in the electro-acoustic system simply by applying to the loudspeaker an amplified version of the pressure measured at the microphone. This is known as the Larsen effect.

In its usual form the Larsen effect consists of a positive feedback loop through a chain of elements that includes the electric and acoustic elements. The latter can be thought of a simple delay, because the acoustic wave travels in a single direction. The case of the transmission through a resonator is slightly more complicated because both the wave traveling from the loudspeaker to the microphone (p^+) and the one in opposite direction (p^-) have to be taken into account [Bailliet (1998)].

The standing wave formed by the superposition of these two waves also affects the performance of the loudspeaker, as will be shown in the next section.

AUTO-OSCILLATION OF A CLOSED-LOOP ELECTROACOUSTIC SYSTEM

In this section we describe the problem of a loudspeaker coupled to a one-dimensional resonator, with the aim of determining the oscillation frequencies of the coupled system.

Description of the system

A loudspeaker driving the acoustic field in a straight resonator is placed at the origin of the reference frame (along the plane $x = 0$). In this plane the pressure P and an acoustic velocity V are coupled to the loudspeaker.

A microphone placed at $x = x_\mu$ converts the acoustic pressure in this plane (supposed to be evenly distributed over the plane) and converts into a voltage u_μ.

The elctrodynamic loudspeaker is represented here by its Thiele-Small model, including its mechanical apparatus – a single mode mass-spring and damper system forced by the electrical force and the acoustic driving of the medium. In sinusoidal regimes the model can be described by a matrix relating the force of membrane on the air (F) and the voltage difference between the terminals of the loudspeaker U to the v displacement velocity of the membrane V, and the electric current flowing through the coil I.

$$\begin{bmatrix} F \\ U \end{bmatrix} = \begin{bmatrix} Bl & Z_m \\ Z_e & Bl \end{bmatrix} \begin{bmatrix} V \\ I \end{bmatrix} \qquad (1)$$

where l is the length of the coil immersed in a magnetic field of intensity B. This model is only valid for small displacements of the membrane. The non-homogeneity of the field $\mathbf{B}$, for instance is one of the sources of non-linearity in the loudspeaker. We will restrict our study to the linear range of the loudspeaker.

The mechanical impedance is a characteristics of the loudspeaker, and it corresponds to the impedance seen by the electric motor in vacuum:

$$Z_m = \frac{F}{V} = j\omega M_m + R_m + \frac{1}{j\omega C_m} \qquad (2)$$

When immersed in air, F acts on the acoustic medium with impedance Z_{ac} to produce a displacement velocity V. For the current work it will be sufficient to consider the front impedance which has bigger magnitude variations, of a factor of more than 1000. The input impedance of the bore can be written as $Z_{ac} = \frac{PS_l}{Q/S_l}$, where S_l is the effective area of the loudspeaker membrane and PS_l and Q/S_l are respectively the force and flow imposed to the air.

Using Z_{ac} in equation (1) we can derive a transfer function between the voltage and the acoustic pressure ($P_0 = F/S_l$) at the surface of the loudspeaker:

$$U = \frac{1}{Bl}\left(Z_e - \frac{Z_e Z_m + (Bl)^2}{Z_{ac}}\right) P_0 S_l \qquad (3)$$

In the complete electroacoustic system, the loudspeaker input U is eventually the result of the measurement of the pressure at the position of the microphone P_μ, again supposed to be related linearly to U. $H(\omega) = \frac{P_0}{P_\mu}$ is the transfer function of the open-loop system:

$$H(\omega) = \alpha \frac{1}{Bl}\left(Z_e(\omega) - \frac{Z_e(\omega) Z_m(\omega) + (Bl)^2}{Z_{ac}(\omega)}\right) \qquad (4)$$

where we defined a parameter α independent of frequency combining several factors that affect the gain, in particular the microphone sensitivity, the amplification and effective cross-section of the resonator.

Criteria for auto-oscillation

The closed-loop system occurs once the microphone is coupled to the acoustic resonator and is in its turn influenced by U_0. In this case, a feed-back loop is established with its own oscillation frequency. The Nyquist stability criteria provide a simple method to calculate it. The auto-oscillations are related to unstable regimes of the system, i. e., for which a perturbationin the resonator pressure grows exponentially. Not only a positive gain is necessary, but also should the perturbation arrive in phase after a complete loop. We thus seek the condition:

$$|H(\omega)| > 1, \qquad \arg(H(\omega)) = 0 \qquad (5)$$

In order to get some insight on the working frequencies of the system we can analyse the magnitude of the terms in equation (4) for the minima and maxima of the impedance of the resonator (figure 1), and compare to the working frequencies of a clarinet.

Clarinets, like other reed instruments work on maxima of the impedance of the resonator Z_{ac}. In such instruments, the first extrema of Z_{ac} have a considerably big dynamics, maxima and minima being often 60 dB apart.

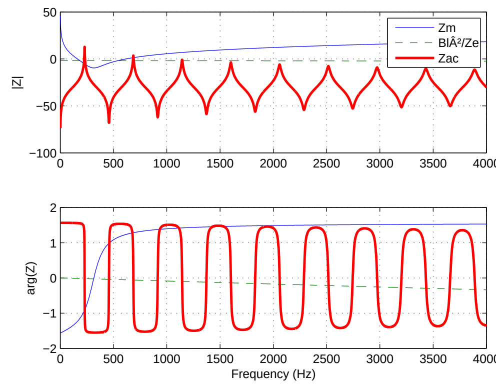

Figure 1: Comparison of the terms intervening in the transfer function (eq. 4)

For the electro-acoustic clarinet used in the experiments that follow, $\frac{(Bl)^2}{Z_e}$ does not vary considerably in phase for the first few resonances. The first few zero-phase values of H are closely related to the resonance frequencies of the bore. The phase of the transfer function then follows $arg(Z_{ac}) - arg(Z_m)$. The mechanical impedance of the loudspeaker shifts the overall phase of the transfer function to the negative imaginary plane. Due to the losses in the resonator, eventually the phase of H stops crossing the real axis.

Several frequencies satisfy the zero-phase criteria. The actual minima around which the auto-oscillation is established depends both on the relative gains $|H|$ at the points of zero phase of H but also on the initial perturbation of pressure or voltage in the circuit (described below).

An interesting remark about the conditions described above is that when the polarity of the loudspeaker is inverted, the argument of H is opposite to the one described above. The auto-oscillation happens when the phase crosses $\arg(H) = \pi$. In the example given in figure 2 this does not happen. However when taking into account the propagation between the loudspeaker and the microphone, the transfer function crosses the negative real semi-axis in two points close to the resonance, but that move further from it as the microphone is moved away from

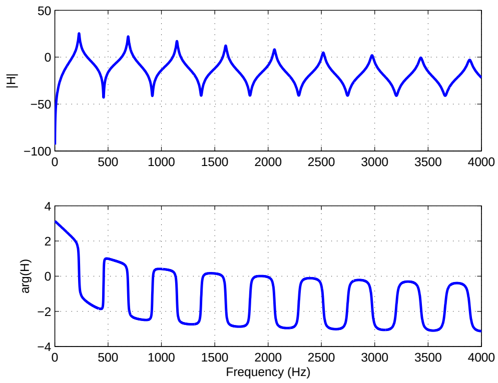

Figure 2: Plot of the modeled transfer function for parameters measured in the electroacoustic system

the loudspeaker. Crossings of the π-phase occur at lower $|H|$ values than the 0-phase crossings.

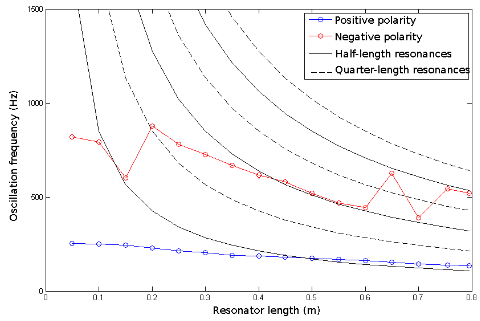

Figure 3: Measured frequencies of oscillation of the closed-loop system for different resonator frequencies and comparison to the modes of the resonator

In the real system, the oscillating frequencies were measured for different lengths of the resonator. As can be seen, with inverse polarity these are quite different from the expected quarter-length resonances of the bore. Moreover, the gain had to be considerably increased relative to the one used in positive polarity. In positive-polarity mode, the measurements follow roughly the quarter-wave length resonance, although for short bore lengths, it tends to stabilise near the resonance of the loudspeaker (~400Hz).

TRANSIENT REGIMES

As described above, the electro-acoustic “woodwind” can be considered linear at low amplitudes. As such, a small perturbation is expected to start with an exponential growth. The same can be said of an instrument such as the clarinet, for which small oscillation can be considered to take place in a linear region of a strongly non-linear characteristic curve [Kergomard (1995)].

However, as mentioned in the previous section, the actual regime the oscillation starts on can depend on the details of the perturbation used to initiate the oscillations. In this section we analyse the influence of the spectral characteristics of the perturbation.

As a test for this hypothesis, a simple experiment was developed, consisting on the investigation on the influence of the spectral content of an artificially injected perturbation in the electro-acoustic system.

A summation circuit is used to artificially add a controlled perturbation to the the sound emitted by the loudspeaker. The injected signal is added to the regular feedback loop. Signals used for this experiment consist of sinusoidal bursts with different periods and amplitudes and repeated every 0.5 seconds. The single period burst has a spectral content distributed around the frequency of the sinusoid (and odd multiples) and tends to zero at $2n$ times this frequency.

To prevent an auto-oscillation to be created spontaneously before it is initiated, the circuit is closed during the recording of the experiment, while the burst is being injected. Figure 4 shows an acquisition of the resonator pressure (in light gray) and the injected burst (dark).

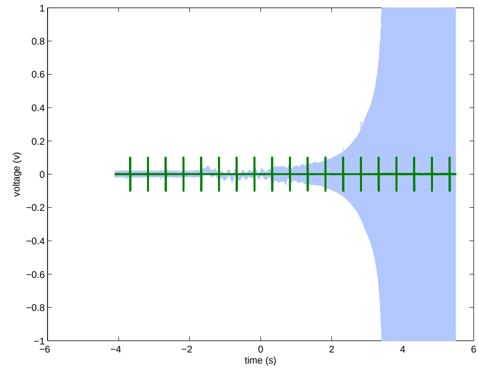

Figure 4: Resonator pressure and sinusoidal perturbation burst used to initiate the oscillation

Observations

As an example, two observations for different frequencies are compared here. The first, at a frequency of 220 Hz, is unrelated to the first frequency of auto-oscillation of the system, in this case 470 Hz (fig. 5). The logarithmic plot shows a phase of exponential growth before reaching a short faster phase, then followed by saturation. A slight influence of the bursts can be seen every 500 ms while the oscillation has low amplitude.

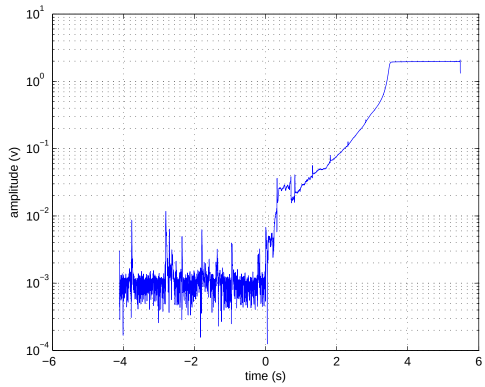

Figure 5: Envelope of the oscillation for a perturabation spectrally unrelated to the resonance frequency

Another example (figure 6) shows a similar experiment, with similar amplitude for the injected bursts, but now at a frequency of 440 Hz, close to the oscillation frequency of the system. The oscillation is seen to grow from a much larger amplitude as soon as the first burst arrives. The burst is not seen as a peak as before, but as a step, since most of its amplitude is immediately injected in the oscillation. A negative step can be seen before saturation, probably corresponding to a negative phase of the burst relative to the oscillation.

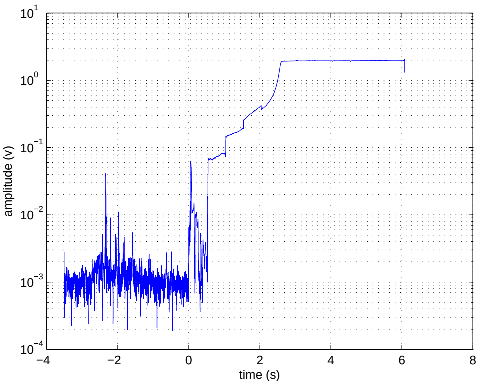

Figure 6: Envelope of the oscillation for a perturabation spectrally related to the resonance frequency

Although in the first example the oscillation is also seen to grow, this is probably due to the injection of amplitude by the background noise (wide-band). The exponential growth starts at a lower amplitude and the successive perturbations injected do not affect the growth of the initial perturbation.

Similar remarks can be made in the attack of a musical note in a clarinet. Although the sinusoidal burst is not a realistic attack transient for the mouth pressure for instance, linearly increasing ramps with different slopes can inject energies with different spectral distributions. A matching of the ramp speed and auto-oscillation frequency can favour a quicker growth of the note in an instrument.

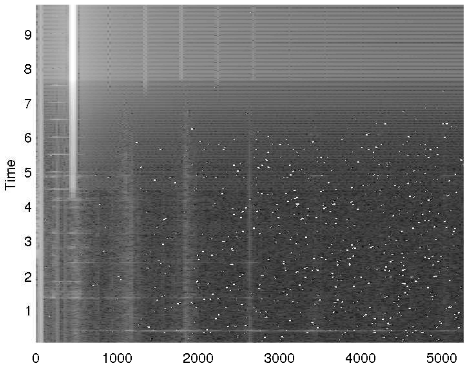

Figure 7: Spectrogram of the oscillation corresponding to figure 5. The periodic bursts can be seen after 5 s as a horizontal line with maximum amplitude at 200 Hz. Also notice the slight misalignment between the auto-oscillation and the nearest mode of the resonator (brighter vertical shade in the background noise)

CONCLUSION

The electro-acoustical woodwind can be an interesting model system on which to study the modes of oscillation and transient phenomena occurring in woodwinds, mainly because the input parameters related to the mouthpiece can be precisely controlled. The work presented here is a first study characterising such a system. Some complications were identified in this study, in particular the fact that the oscillation frequencies are affected by the characteristics of the loudspeaker. A complete emulation of the musical instrument will require, other than the shaping of a non-linear relation between the input and output pressures, a circuit to balance the effect of the microphone / loudspeaker configuration.

Nevertheless, even in a mostly linear system, it was possible to demonstrate that the initial growth of the perturbation is dependent on the relation between its frequency content, and the oscillation modes of the electro-acoustic system. Further studies of the transient will be performed in more controlled noise conditions, and different perturbation shapes. These will be related to recent studies in numerical simulations [Silva (2009)].

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