Computer Aided Measurement and Theoretical Models of Interaction of Musicians and Brass Instruments

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COMPUTER AIDED MEASUREMENT AND THEORETICAL MODELS OF INTERACTION OF MUSICIANS AND BRASS INSTRUMENTS.

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SUMMARY

In brass instruments the interaction between player and instrument ist very strong and this fact tends above all others to confuse most experiments. To get more information about the structure of this interaction different measuring methodes are necessary or can be substituted by transforming the obtained data from frequency domain into time domain. For simulation, models based on electrical circuits can be used. This paper gives a brief report on the current work .

0 INTRODUCTION

According to the playing experience of brass players the quality of a brass wind instrument can be defined by

a) internal intonation
b) response
c) sound quality

These terms belong to the terminology of musicians and there is no simple relationship to physical parameters of the instrument. An idea of the complex relationship between musical parameters, mechanical parameters of the instrument and measured physical parameters gives figure 1.

Whereas the information about internal intonation errors can be obtained from an input impedance measurement with the hard- and softwaresystem BIAS, which was presented the first time at the 87th AES Convention in New York [1], the judgement of response and -partly- the sound quality needs measurement methodes which operate in both frequency <u>and</u> time domain.

Therefore a model of the interaction between the player and the instrument can be useful, which provides the simulation of the transient and nonlinear behavior of the whole system player-instrument. As a lot of electrical simulation software packages based on SPICE-models appeared during the last years, we tried to design an electrical model consisting of two parts:

a nonlinear oscillator circuit (player) and
a linear load (instrument).

The electrical model of the mechanical oscillator is chosen as simple as possible. Only the values of two parameters are important and can be changed:

the air pressure of the lungs and
the tension of the lips.

Under the assumption that the behavior of the instrument is linear [2], it is possible to transform the frequency domain data of real instruments obtained by BIAS, into the time domain using theortical and mathematical methods. This allows now the simulation of the whole system in the time domain, considering the load-feedback to the stimulating oscillator to calculate the overall system response.

1 Measurement Arrangement - HARDWARE

For our investigations we use an improved version of BIAS, which performs an input impedance measurement inside the mouthpiece exactly in the plane of the lips. Therefore the obtained data represent the enviroment, which is given for the lips of the player during playing his instrument. Detailed information on the physics of brass wind instruments and the principles of acoustic impedance measurement are given in [3,4,5].

1.2 Hardware components

Based on the above mentioned system the improved hardware consists of:

- a computer controlled digital sine wave generator B&K 1049
- an artificial voice B&K 4219
- two identical measuring microphones Neutrik 3382

- a new developed modular computer controlled amplifier and analyzer module "WinAlyze"
- a IBM-PC compatibleComputer with a numerical coprocessor
- a DASH-16 data acquisition board and
- a GPIB controller.

A block diagramm of the arrangement shows figure 2.

The features of the computer controlled amplifier and analyzer module WinAlyze (figure 3) are:

- A power amplifer, which amplifies the sinusoidal generator signal. The available AF power is up to 50 W rms in a 4 Ohm load with neglecting distortion products. The power stage is able to drive the B&K 4219, or commonly used loudspeakers to realize more powerful artificial voice systems.

- Two identical microphon amplifiers, which offer gains from 0 to 60 dB. The gain setting can be done manually, or controlled by the I²C-Bus interface [6].

- In the analyzer part the signal of each microphon channel is rectified and filtered. A phase comparator is built in for future applications. All the analog signals are send to an interface, which is compatible to the DASH16 board inside the PC, including the I²C communication.

The B&K 1049 Sine and Noise Generator is connected to the PC by the GPIB interface. The output amplitude, frequency and the parameters for the compresser loop are set under software control. The generator signal is amplified by the WinAlyze modul and feed to the loudspeaker (artificial voice B&K 4219). The amplified signal of the build-in-reference (condenser) microphone feeds the compressor loop to stabilize the sound pressure constant the frequency sweep.

The signal of the measuring microphone inside the mouth piece is preamplified by the WinAlyze modul, where individual amplification settings under software control can be done. After preprocessing by rectification and filtering, this signal is send to one of the analog input channels of the DASH16 board inside the PC .

2 Measurement arrangement - SOFTWARE

Beside the kernel operations like data acquisition, storing, evaluation etc., a lot of different hardware controlling and setting work has to be done. The software package ASYST was chosen, because it offered easy procedures for controlling GPIB interface boards, performing data acquisition with various bords e.g. DASH16, evaluating data with complex numerical utilities and presenting the results in graphic modes. Since this package is not a brandnew one and the upgrades don´t offer modern memory management, a lot of work around was necessary to process the large numbers of data points.

The test software uses hirarchic organized menues for the selection of measurement procedures, data store and display, intonation error calculation and hardcopies of the plots.

2.1 The measurement kernel procedure

The acoustical impedance is measured within a frequency range from 20 to 5000 Hz in steps of 1 Hz. Start and stop frequency can be chosen within this band. The frequency control of the B&K 1049 is done via GPIB commands in realtime, each value of the current test frequency is set step by step. Due to this fact no synchronisation problems between supposed and real test frequency exist. To avoid test signal distortion the slew rate of the compressor loop must be set to proper low values [7]. This is done automatically according to the actual test frequency. The compression rate is checked to ensure settled and constant test signal amplitude.

Near to acoustical resonances of the musical instrument, the settling time of the instrument increases. For this reason, the frequency stepping time has to be changed by the software. To avoid measurement artefacts, the test procedure waits after each frequency step for a stabilized response amplitude. The mean value of ten samples is built and the deviation of each sample to the median value has to be less then a certain value, which we call "noise band". In the case of a noisy or less constant signal, further software filtering starts, controlled with a parameter called "repetition rate". This alogorithm solves also proplems at low test frequencies, were the signal ripple increases because of the fixed filter cut off frequency after rectification.

2.2 Data storing and identification system

Analyzing a brass wind instument, a lot of measurement procedures are done, caused by different valve combinations and other parameters. Therefore the data file of each impedance plot is provided with a special file extension, where the instrument type, the valve combination and processing information is coded. Individual comments and a short information about actual hardware parameter settings are tagged to the data file.

2.3 Calculation of intonation error

Players usually are able to produce a musical tone only at, or near the resonance frequencies of the brass wind instrument. These resonance frequencies can be found as local maxima values in the acustical impedance plot. Using particulary interpolating algorithms, it is possible to estimate this playalble tones at a higher resolution than the measurement frequency steps of 1 Hz. This is a very important feature for low frequencies. A one Hertz aberration at the lowest natural tone, typically about 44 Hz (for horns) would cause an intonation error of more than 39 cent (or a third of a semitone)!

For the judgement of intonation the frequencies used by musicians producing musical tones are compared with corresponding frequencies of the tempered tonal system. As a player usually adapts his instrument due to a special chamber pitch (a1 = 440 Hz - 448 [!] Hz) or to the intonation of other instruments, and the measurement takes place with a certain position of the tuning slide, the algorithm has to find out the "individual" reference pitch of the instrument with the best "over all" intonation.

A reliable algorithm to solve this minimum problem has to detect the instrument type, valve combination and the ordinal index of the local peaks. According to the fact, that the frequency intervall of two natural tones (resonance peaks) gets smaller in the higher range of the Instrument (according to the system of tempered scale; only few resonance peaks are then really used for producing musical notes), it seems to be a good decision to start analyzing at low frequencies in spite of usually big errors of the first natural tone. In an iterative process, the lowest frequency is analyzed and a temporary ptich is derivated. After this, together with the frequency of the next higher peak, a new temporary

ptich with minimum error is derivated and so on. After all, the last pitch becomes the reference pitch and the particular intonation errors for all natural tones can be calculated and redefined. Figure 4 shows such a plot for a french horn.

3 Frequency domain versus time domain - some considerations.

Input impedance versus frequency allows a relative simple handling of data for calculating intonation errors. The procedure becomes rather complex, if exact accordance to playing experience is required, but nevertheless this method is useful for intonation judgement. Best accuracy is reached because of settled signals and in time correction of the actuator error through the compressor loop. Even actuator errors of more then 50 dB can be compensated without loosing signal to noise ratio during data acquisition.

A disadvantage of this test method is the consumption of a lot of time, stepping through the chosen frequency range.

Using impedance plots no direct visible information on settling time of the instrument, its response or effects caused by unwanted reflections inside the tube is available. But this phenonemas are an important part of the acoustical behavior of an instrument and influence the "feeling" of the musician.

To analyze such phenomenas, Dirac-pulses are required. Taking in account, that such pulses are not realizable, the consideration of system response on signals like short pulses or sine-bursts is a common used test method. Real stimulation signals like those pulses represent only an approximation of an ideal short pulse, various steps of postprocesssing (deconvolution) are necessary.

System theory offers well defined relationships between frequency and time domain characteristics for linear systems [8]. The frequency spectra of a time function can be obtained using Fourier or Laplace transform. Since data usually exist in digital format, various well known numerical algorithms e.g. the FFT algorithm can be used.

Using the transform and the invers transform algorithm, the switch between time and frequency domain is possible.

3.1 The frequency domain

A linear time invariant system is usually described as

$$S2(f) = TF(f) \,.\, S1(f) \qquad (1)$$

where S1(f) is the input singnal, TF(f) the frequency transfer characteristic and S2(f) is the signal response.

In our case we assume a model, similar to the wave parameter.

$$p(f) = Z(f) \,.\, v(f) \qquad (2)$$

The equation (2) can be understood in the following way: resulting sound pressure (p) is a reaction on a certain volume velocity (v), depending on the impedance charcteristic of a passive system (Z) - the instrument. Linearity as a presumption of using signal theory can be assumed up to sound pressures of 130 dB SPL, where the influence of thermodynamics can cause distortions [2].

The terms of equation (1) and (2) include phase information which are represented by the imaginary part of the complex values. However phase information cannot be obtained from our measurement arrangement, we are forced to calculate phase relation by suitable methods (Hilbert transform [8]).

3.2 The time domain

According to the equation (2) a linear system is described as

$$s1 = g * s2$$

whereby * denotes the convolution operation, s1 and s2 represents adjacent signals.

$$\begin{array}{lcl} s1(t) & \circ\!\!-\!\!\bullet & S1(f) \\ s2(t) & \circ\!\!-\!\!\bullet & S2(f) \\ g(t) & \circ\!\!-\!\!\bullet & G(f) \end{array} \qquad (3)$$

s1(t) and s2(t) are the signals in the time domain. g(t) is usually named weighting function.

The corresponding functions can be obtained by the Fourier and inverse Fourier transform

$$\begin{aligned} G(f) &= F \{ g(t) \} \\ g(t) &= F^{-1} \{ G(f) \} \end{aligned} \qquad (4)$$

The Dirac impuls d(t) is the most interesting theoretical test signal in the time domain because of

$$s2(t) = g(t) * c\, d(t) = c\, g(t) \qquad (5)$$

With equation (5) we get the weight function g(t) at once as the response s2(t), known as the "system response". Thereby c is a constant value, delivering the correct physical dimension.

3.3 Fast Fourier Transform (FFT) and inverse FFT

The Fourier transform is definded by the integral of a function in mathematical sense. For application on digital data, discrete fourier algorithms (DFT) have to be used, which are valid only for time and frequency band limited functions. FFT, a very fast algorithm has the only disadvantage to work only with data arrays length within a power of 2. At this point we should remarque, that the result of such a transform can be complex even operating with a source consisting of "real" input data.

To readjust the data array size, the input data are extrapolated to the new margin. Input data in the frequency domain have to be supplied with a negative frequency part, whereby the magnitude is an even and the phase an odd function depending on frequency. Therefore noncomplex output data (time domain) can be generated from complex input data (frequency domain).

3.4 Hilbert Transform

Acoustical impedance measurement on brass instruments is usually done without any observation of phase relations. Figure 5 shows the result for a "F-Horn" with a tube length of about 3,6 m. Realizing a (acoustical) velocity source, by using a tube (length 0 6 cm) filled with thinn steal wires, causes a more or less unknown time delay at the output plane, in addition the simple measurement of the proper phase reference is in practice difficult. The data contain only magnitude information of the complex imdedance values.

Control theory delivers the relationship between magnitude and phase of an realistic system with the help of the Hilbert transform [8,9]:

A realistic (causal) signal s(t) is defined: for $t > 0$
for $t < 0$ $s(t) = 0$

$$s(t) \circ\!\!-\!\!\bullet S(f) = A(f) + i\,B(f) \qquad (6)$$

where

$$B(f) = H\{\,A(f)\,\} \qquad (7)$$

The Hilbert transform is defined as

$$f_2(x) = H\{f_1(x)\} = \frac{1}{\pi}\int_{-\infty}^{+\infty} \frac{f_1(x)}{x-y}\,dy \qquad (8)$$

The numercal computation of these integral leads to useless results because of two problems:

1. the infinite integration limits and
2. the formula causes an error (pole), when $y = x$ [10]

Our approach to avoid the above mentioned problems is:

Implementation of a numerical method for Hilbert transform using FFT algorithms [11,12]. Hence the Hilbert transform of a signal causes a 90 degree phase shift in its spectral function, a data set has to be processed with FFT. After adding the 90 degree values a inverse transform takes place. Figure 6 shows the calculated phase relation for the impedance plot of the F-Horn of figure 5.

A synthetic system response derivated with this method from the impedance plot of the instrument above is shown in figure 7. The plot illustrates the applicability of the used method. The calculated delay of the response matches exactly the time needed by the wavefront travelling from the mouthpiece to the bell and back.

4 Some results of a first attemt on simulation

The basic idea is to create a behavioral model in terms of electrical circuit elements to approximate the physical scenario producing a "musical" tone on the brass instrument.

In physical terms energy is delivered by the pressure of the lungs which activates the mechanical resonator with a valve function. The actual cross section area modulates an air flow streaming into the instrument. The controlling force of the valve is defined by the difference of pressure between mouth cave and mouth piece. This means in electrical terms:

- a voltage source with a particular source resistance
- a capacitor representing the volume of the mouth cave
- a controlled admittance (valve function)
- a transmission line model as instrument

As showen in figure 8 the voltage difference between load and source activates the LC resonator, controlling the admittance. A particular voltage is necessary for initialize the conduction. The whole system oscillator - load represents now a "labile" system where the load does not act like a simple filter, but influences the signal-shape during generation.

Figure 9, 10, 11, 12 and 13 illustrate some results of this simulation.

Acknowledgements

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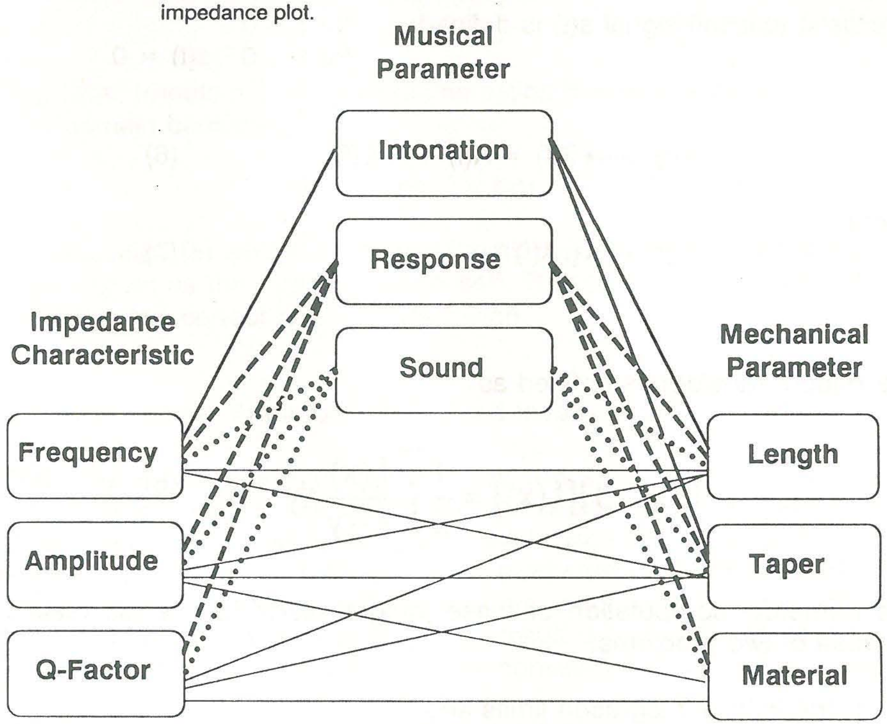
Fig.1: Relationship between "musical" parameters, mechanical parameters of the instrument and characteristics of the acoustic impedance plot.



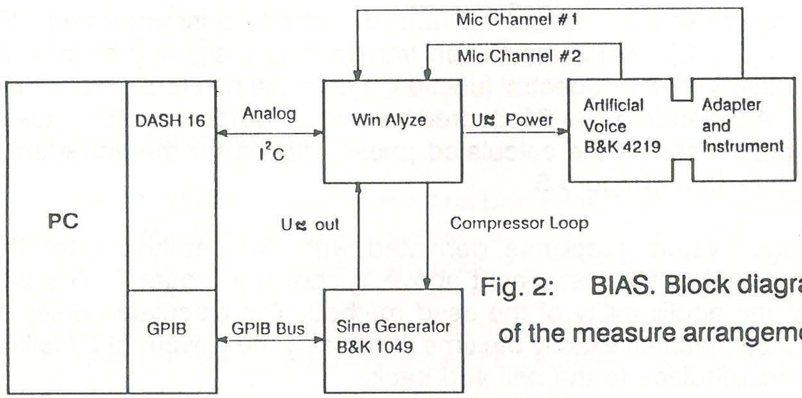


Fig. 2: BIAS. Block diagram of the measure arrangement.

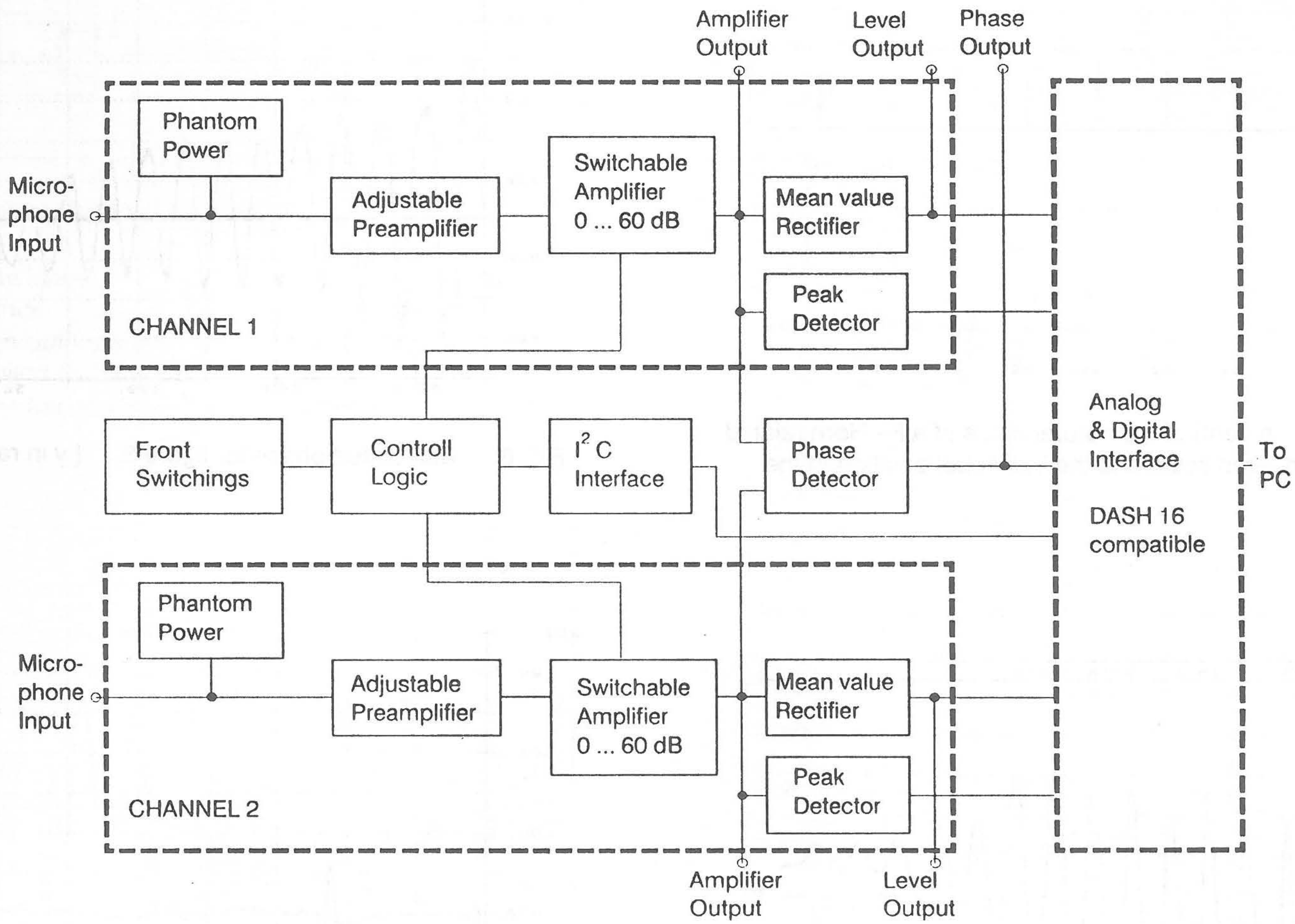


Fig. 3: Block diagram of the computer controlled amplifier and analyzer module WinAlyze.

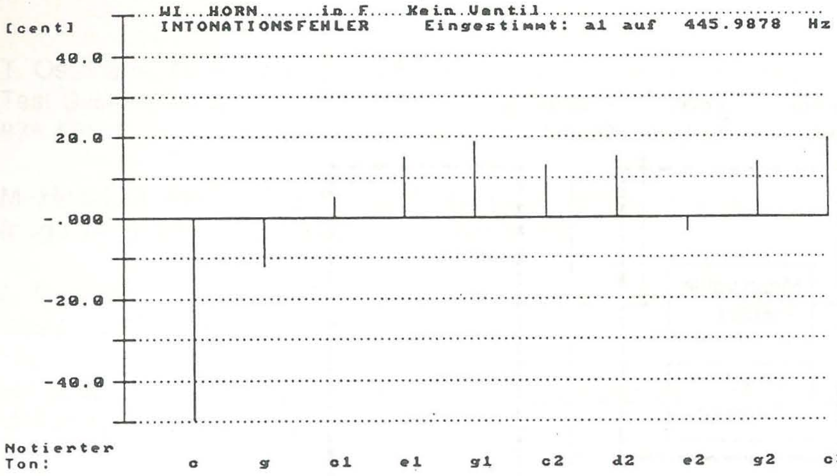


Fig. 4: Intonation error (in cent) of the natural tones of a F - Horn related to the equal tempered scale and the individual position of the tuning slide.

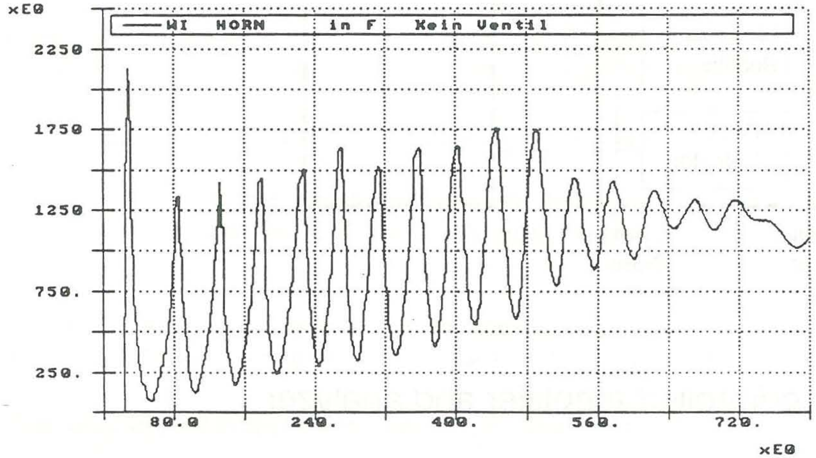


Fig. 5: Impedance plot of a f - Horn. The peaks indicate the position of the playable natural tones on the frequency axis.
[x in Hz, y in AU]

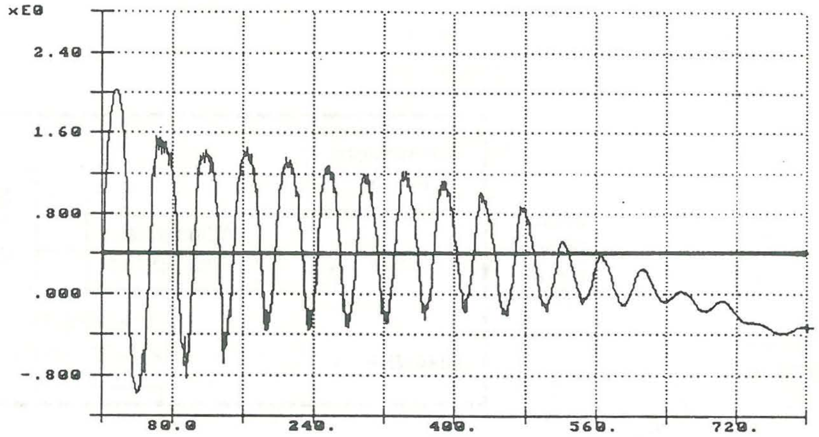


Fig. 6: Calculated phase for figure 5. [y in rad, x in Hz]

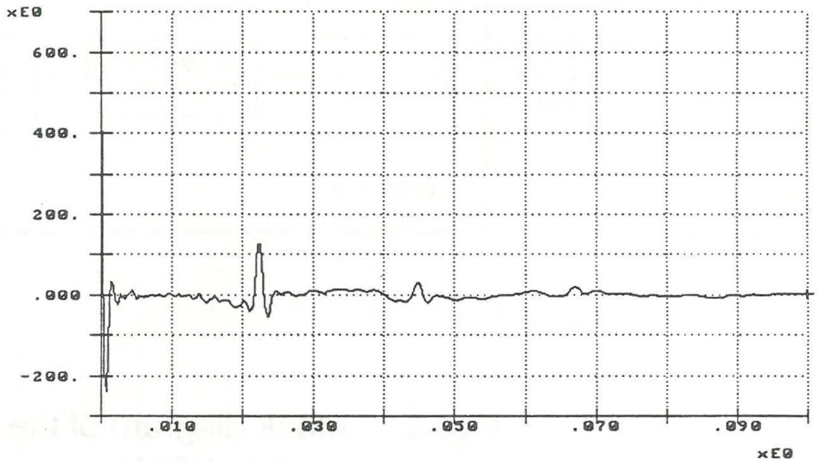


Fig. 7: Calculated impulse response for the F - Horn.
[y = relative air pressure, x = sec.]

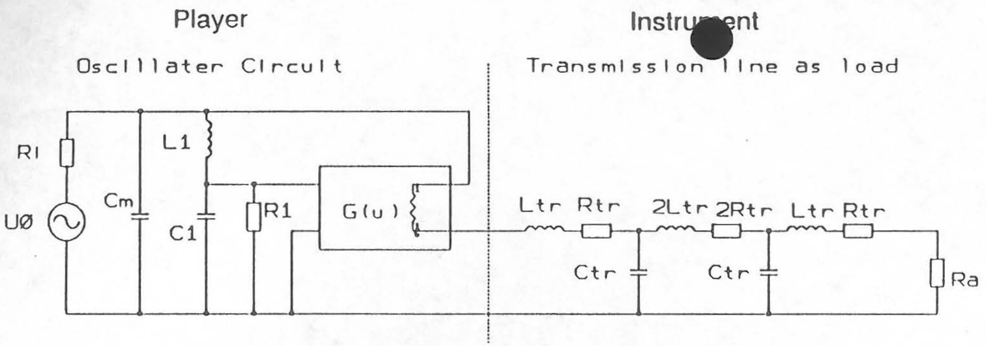


Fig. 8: Simulation of the whole system in time domain.

U 0 = pressure of the lungs
Ri = source resistance
Cm = volume of the mouth capacity
G(u) = Voltage controlled admittance
*tr = Elements of the transmission line (instrument)

C1 = lip tension
L1 = lip mass
Ri = loss

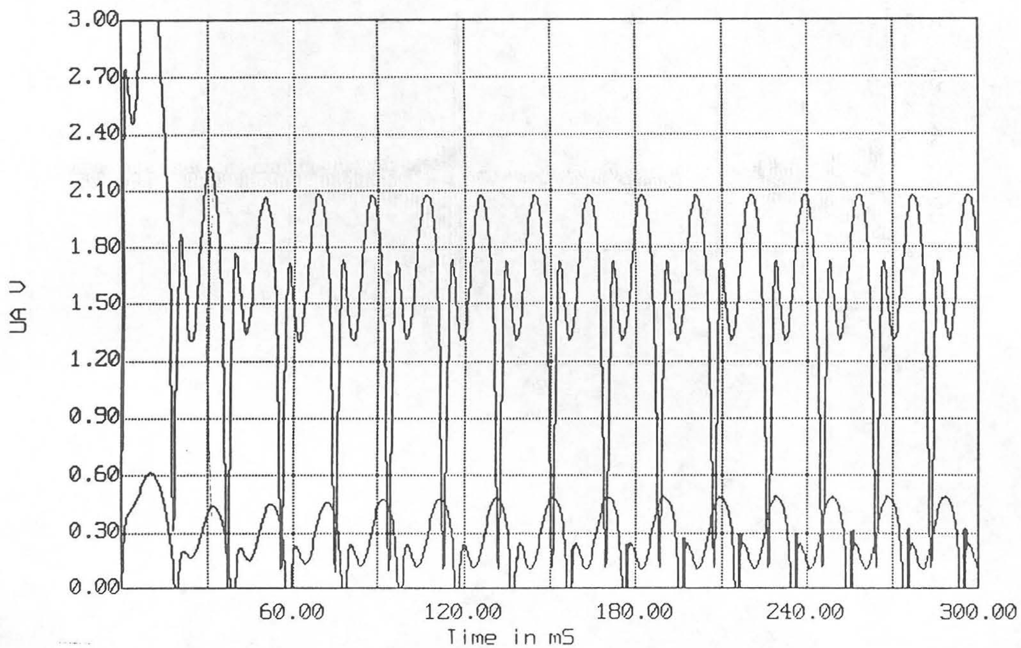


Fig. 9: Simulation. Relationship between air pressure of the lungs and frequency of the regime of standing waves inside the horn. A shifted up air pressure leads to a higher "playing frequency" and vice versa. Note the different waveform depending on air pressure.

Fig. 10: Simulation. A player tries to blow the note 10% lower than the correct excitation frequency.

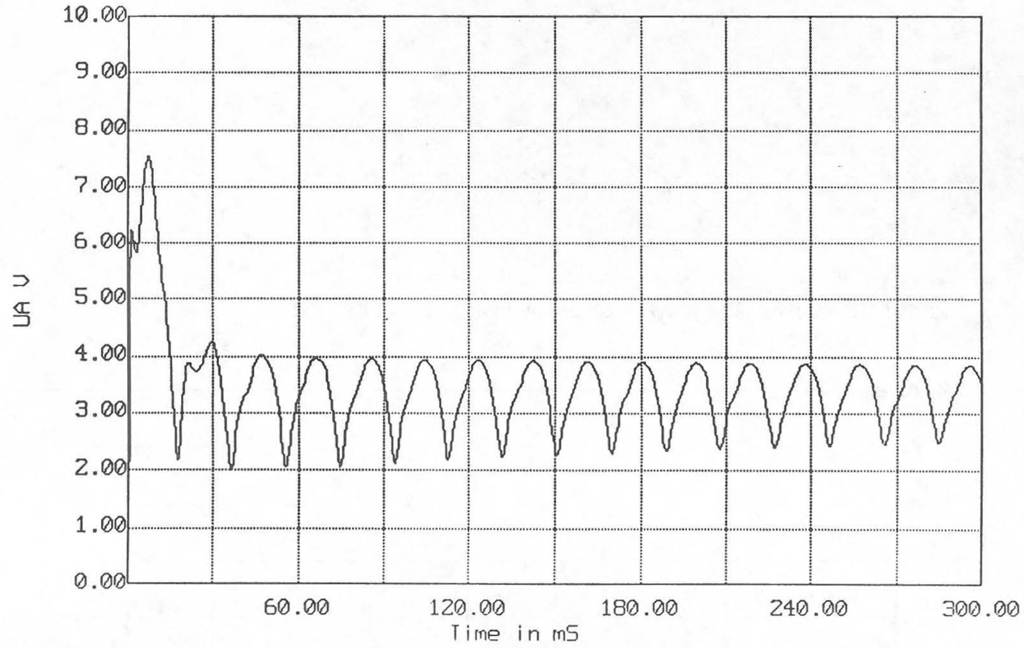


Fig. 11: Simulation. A player tries to blow the note 10% higher than the correct excitation frequency is.

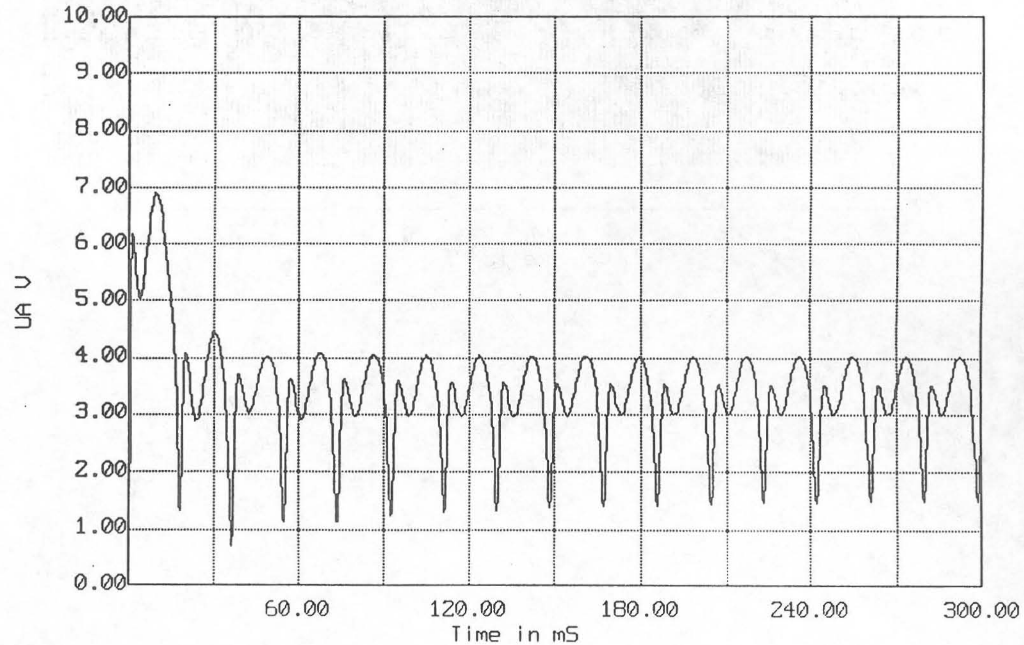


Fig. 12: Simulation. The player excites the instrument at the correct frequency.

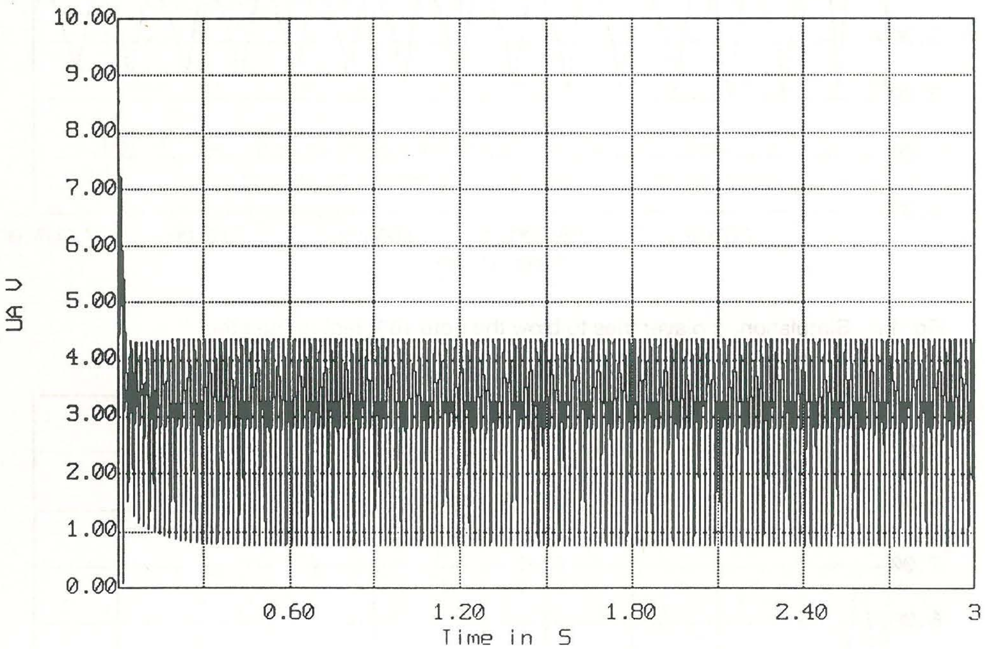


Fig. 13: Simulation. Time function decay of the incorrect "blowen" tones of figure 10 and 11.