A MODULAR APPROACH TO EXCITATOR-RESONATOR INTERACTION IN PHYSICAL MODELS SYNTHESIS

G. Borin, G. De Poli, A. Sarti
C.S.C.-D.E.I. University of Padova, Italy

Abstract: A proposal of a general and modular scheme for musical instrument synthesis is presented. A complete example is illustrated, showing a model of hammer-string interaction in piano sound.

Sound synthesis by physical models, thought not recent, has gained interest during last years in computer music field, for the natural quality of sound it exhibits in relation with the decreasing computational cost. The different approaches and the great variety of proposed solutions for the realization both of specific intruments and of general structures, requires the developement of a unifying conceptional framework and a coherent metodology. We belive that in computer music the main aim of physical model synthesis is not the simulation of a specific traditional instrument, but the construction of tools, which may allow a great deal of timbral choices and a flexible and simple use. Each musician should be able to create his own instruments, even new and unheared ones, without loosing contact with his acoustical experience and intuition. This requires the developement of a descriptive method of great generality and modularity.

In this work we expose a general approach for modeling musical instruments. This method was originally conceived as an aid for the realization of a model of the hammer-string interaction in piano sound. The pourpose was to find a conceptual structure in order to assemble known models of excitators and resonators to obtain new synthesis models, and to compare their effectiveness. The main idea is that in every natural musical instrument, in the wider extension of this term, it is possible to recognize a common basic scheme: an excitator and a resonator which interact. This is the framework we start from in order to develop simple and complex models. The resulting scheme allows to develop excitator and resonator in a fairly independent manner, thus giving the possibility of using third-party work in one's own models almost in a "black box" style.

1. Modular approach to musical instruments modeling.

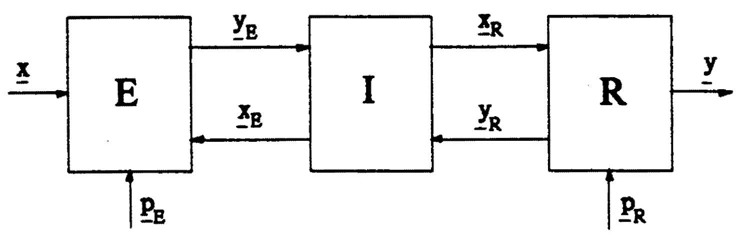
As we told above, we always consider a musical instrument as composed of an excitator and a resonator. Note that these blocks have a functional meaning, and do not necessarily describe physical parts of the instrument. In this work we consider the excitator as the element which causes and, possibly, sustains the vibrations in the resonator. The resonator describes the place in which interesting vibratory phenomena arise. This means that we consider as parts of the resonator both the oscillations supports, such as strings and acoustic tubes, and the elements which are used to alter the produced sound, as soundboards, wind bells and so on.

In connecting excitator and resonator, several schemes are possible. Perhaps the simplest structure is the "feed-forward" scheme, where the excitator acts on the resonator without accepting informations from it. An example of this connection is given in the "initial condition technique", in which the resonator is set by the "excitator" to its initial condition and then is left free to evolve, like in the plucked string in [1]. Another example is the model "source-linear filter" used in speech synthesis field.

A more general and accurate structure is given by the "feedback" scheme [2]. In this scheme the excitation takes into account the state of the resonator, so that a full interchange of informations between the two blocks is accomplished. Lots of applications are possible for this scheme, due of its generality and completeness. Anyhow, it should be pointed out that, in its original form, this scheme requires the direct connection of excitator and resonator. Thus, it is not possible to develop these two blocks independently, having each of them to consider input and output constraints imposed by the other. In the developement of a synthesis structure this can be a serious limitation: a change in the excitator, for example, will probably reflect on the whole structure of the model. Furthermore it is not possible to build a model, coupling excitator and resonator choosen between the ones available in literature.

The structure we propose to overcome these problems, is based on a three block scheme, as illustrated in fig. 1. E and R blocks stand for excitator and resonator, respectively. It is not restrictive to assume that E and R are time-varying dynamic systems, and it is common practice to assume R as linear, while no restrictions are made on E; E and R are so described by means of explicit equations. X is the input vector representing the excitation actions. Pe and Pr are vectors of parameters representing the modulating actions on the instrument. Y is the output vector, representing the musical signals. Other vectors are the information exchanged between E and R through I. Block I acts as an interface block: its main pourpose is to separate block E and R, so that they can be described independently. In order to accomplish this, I block performs an input/output variables adaptation, eηsuring that the output of E is compatible with the input of R, and vice-versa. It should be pointed out that this scheme preserves the feedback structure, and so it has the same description capability of the raw scheme presented above. Furthermore, we note that in this description block I makes blocks E and R act as true "black boxes": in fact a hypotetical user does not need to know the details of the internal structure of E and R, his interest being only in their I/O relations. Finally, many of the problems which interaction description usually brings can be collapsed into block I, and conveniently formalized ad resolved there.

Fig.1



It can be noticed that the interconnection of two discrete systems can give rise to a problem of computability: this happens when the resulting scheme has loops without delay elements. Sometimes an "ad hoc" manipulation of the system equations can eliminate these dependances; generally it is necessary to solve a (often non linear) system for each sample period. An approximate solution, which is acceptable for sufficently high sampling rates, is to insert delay elements in the upmentioned loops; this solution preserves the modularity and gives good computational efficiency. Our three-blocks scheme, having more exchange variable than the two-blocks one, can give rise to these kind of problems more frequently. However, adopting the approximated solution, we can easily overcome these problems without loosing the independency of the blocks.

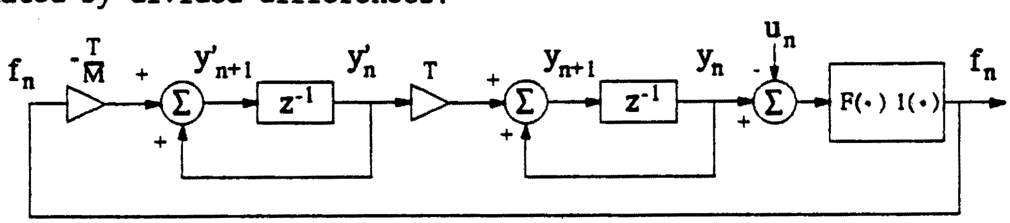
2. Hammer-string interaction in piano sound synthesis.

In this part we expose a complete example of application of the general scheme exposed above. Model realization is based on a work by G. Garnett [3] on piano sound synthesis, in which Waveguide Digital Filters are used for the simulation of strings and soundboard. On the other hand, while in Garnett's model excitation is obtained by mean of a complex and tricky mechanism based on Helmholtz's well known approximation of strucked string, in the model here exposed we use a different excitator, which is based on a accurate -even if simple- physical description of the hammer, and an adeguate interaction mechanism. Moreover, we use a very simplified description for the resonator, our main interest being in the study of the interaction between hammer and string, and not the production of high quality piano tones. We assume that the hammer strikes a single ideal string, connected to a perfectly rigid support by one side, and to a slightly lossy support by the other; this one represents the bridge, and we further suppose to pick up the signal at this point, without sending it to a soundboard model.

Our choice for the hammer model have had two objectives: sufficently good physical description and low computational cost. From this point of view literature reports lots of interesting examples; we choose a proposal by H. Suzuki [4], because of its particular simplicity and good physicity (see also [5] for a more general model discussion). The model we use has a mass, representing the body of the hammer, and a nonlinear spring, with zero lenght at rest, which stands for the felt. Following Suzuki experimental results we assume that the spring characteristic is: $F(h)=[Ah^2+Bh^3+Ch^4]1(h)$ where F [N] is the force applied to the spring, h [m] is the correspondent displacement, A, B, C are suitable constants and 1(h) is the Heavyside function. The hammer motion equation is: $My''=-f$ where $f=F(y-u)$ is the force that hammer exerts on the string, u and y [m] are respectively the position of the string at contact point, and the position of the hammer. Note that the end of the spring which comes in contact with the string has no dynamic by its own, being the spring masseless, and so the contact condition between hammer and string is simply $y<u$, and is realized by the Heavyside function.

Following equations exposed above, the excitator block will provide an input for the displacement information u, and an output for the force f. Excitator model is then easily obtained as in fig. 2, if derivatives are approximated by divided differences.

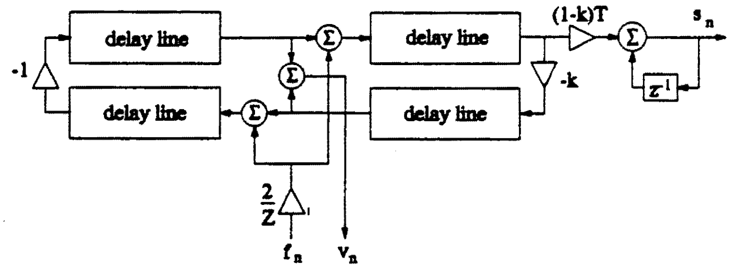
Fig. 2



The resonator model has been realized using Waveguide Filters (WGF) [6]. A WGF is a net in which branches are pairs of digital delay lines, and nodes are passive junctions. Branches propagate undistorted waves in opposite directions, so describing tracts of losseless transmission lines having a constant characteristic impedence. Junctions connect two or more branches, and give account of characteristic impedance discontinuity. Considering the hypothesis given above, the resonator model can be easily realized as in fig. 3, where we assume that signal is a transversal velocity. In the model one end of the string is connected to a junction which operates an almost complete reflection by means of a reflection coefficient K, slightly less than unity. Transmission is given by a coefficient 1-K which weights the velocity signal: digital integration gives the audible signal.

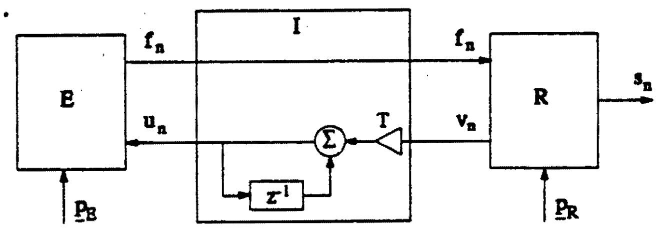
Note that the realization of the interaction mechanism requires a displacement information from the string at contact position; it is also necessary to insert hammer informations in the string. First operation is easily accomplished reading the content of the delay cells in the contact position. The model being linear, perturbation inserting is obtained adding external velocity contributions to the signal already present in the line. Moreover, we notice that the force applied at the contact position sees the parallel of two line tracts of equal characteristic impedance Z: transversal velocity waves are then given by the force signal divided by Z/2.

Fig. 3



If we consider that the contact condition is completely realized in the excitation block, we see that the interaction block has now the only task to adapt exchange variables dimensions. This operation is not necessary for the excitation signal f, which is directly applied to the resonator, while, to obtain the reaction of the resonator, an integration of the velocity signal is requested, as we need position information for the hammer. The resulting scheme is reported in fig. 4; we observe that the interconnection of the blocks does not give computability problems because, even if f is instantaneously dependent from u, yet R has been realized so that it has no instantaneous dependency from v and f. We observe also that resetting the integrator before each hammer hit has no particular consequences, having the only effect of introducing a negligible offset in the mean position of the string, but it has the advantage of avoiding accumulation of integration errors.

Fig. 4



Simulation results, referred to Suzuki parameters [4], are reported in fig. 5, which exposes the case of a quite strong hammer hit (v=3.5 m/s). In this figure three curves are presented. Curve a represents the hammer motion at contact point, curve b the string displacement at contact position and curve c the interaction force which is non-zero only during interaction. A comparison with [4] shows good agreement with diagrams obtained with the same physical parameters, but employing a much more complex model. The course of the corresponding acoustic signal is illustrated in fig. 6. The spectral evolution of the output signal is given for two different hammer velocities in the case of 220 Hz (fig. 7 a,b) and 880 Hz (fig. 7 c,d) strings. For both strings the spectrum widening due to the increase of the hammer velocity is evident. Noisy attack followed by the harmonic sound, is more evident in the higher frequency string due to the longer interaction time compared with the small oscillation period.

References

[1] L. Hiller, P. Ruitz.1971."Synthesizing musical sounds by solving the wave equation for vibrating objects", J.A.E.S, 19(5,6):462-470, 542-551.
[2] M.E. Mc Intyre, R.T. Schumacher, J. Woodhouse.1983. "On the oscillations of musical instruments", J.A.S.A, 74(5):1325-1345.
[3] G. Garnett.1987. "Modelling piano sound using Waveguide Digital Filtering Techniques", Proc. ICMC '87: 89-95.
[4] H. Suzuki.1986."Model analysis of a hammer-string interaction", Proc. ICASSP '86: 1285-1288.
[5] X. Bouttillon.1988."Model for piano hammers: experimental determination and digital simulation", J.A.S.A, 83(2):746-754.
[6] J.O. Smith.1987."Waveguide Filter tutorial", Proc. ICMC '87: 9-16.

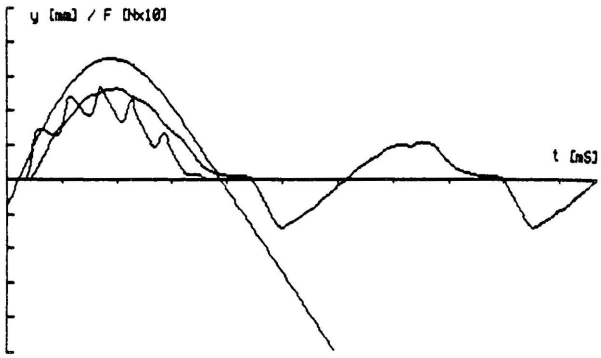


fig. 5

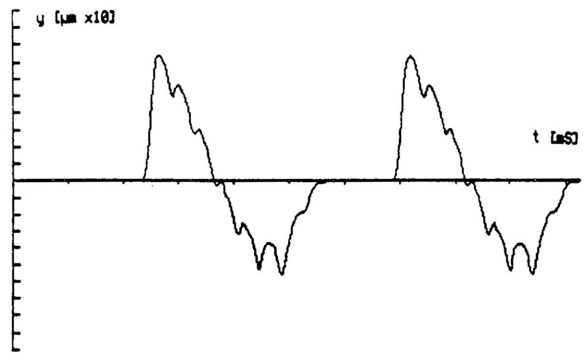


fig. 6.

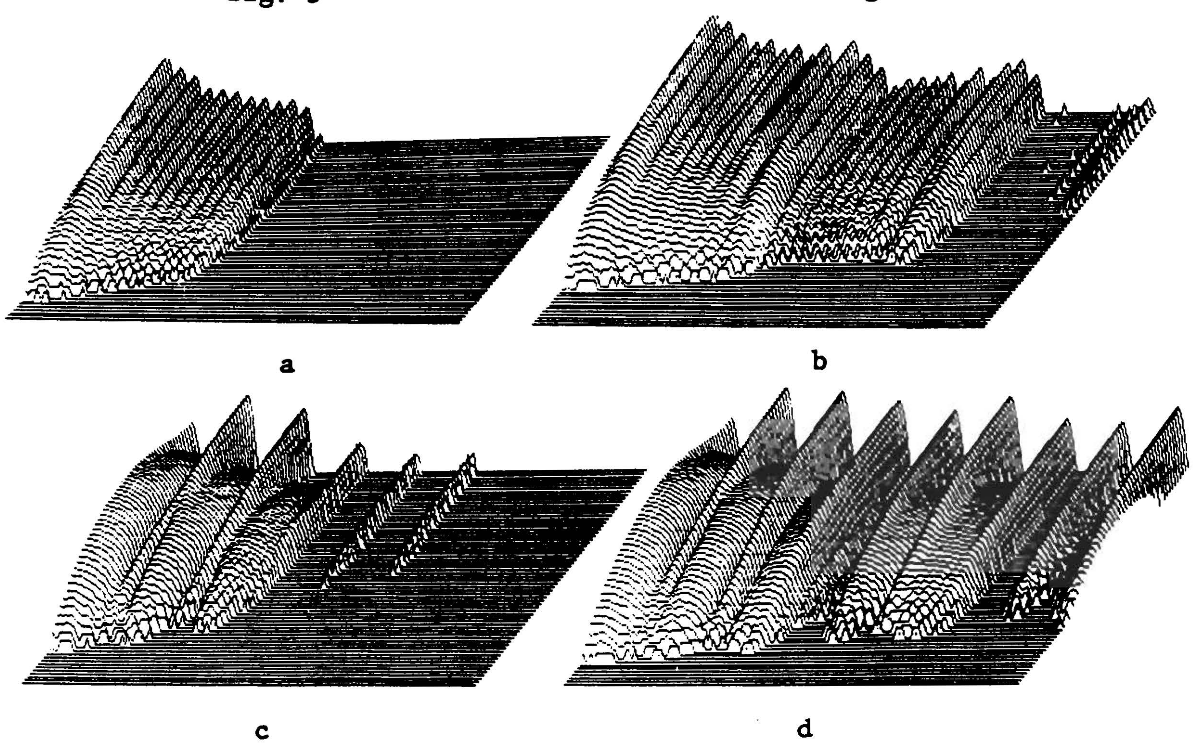


fig. 7