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Source: *Computer Music Journal*, Vol. 16, No. 4 (Winter, 1992), pp. 30-42

Published by: The MIT Press

Stable URL: <http://www.jstor.org/stable/3680467>

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Algorithms and Structures for Synthesis Using Physical Models

Sound synthesis by means of simulated physical models has gained popularity in the last few years. One of the principal reasons for this interest is that this technique, based on modeling the mechanism of production of sound, seems to offer the musician simpler tools for controlling and producing both new and traditional sonorities. In general the aim of any model is to describe the fundamental aspects of the phenomenon in question by means of mathematical relationships. Most often models are used for purposes of analysis. In science and engineering, models are commonly used for the purpose of understanding physical phenomena. This is especially true in musical acoustics, where it is common practice to study a traditional instrument through its physical model in order to understand how it works (Keefe 1992; Woodhouse 1992). In the pioneering work of Hiller and Ruiz (1971), physical models were used with the goal of producing musical sounds. Since that time, physical models have been used for synthesis purposes.

In this article we examine how models can be constructed for musical applications and discuss the principles that inspire the most widely used synthesis algorithms. We will also try to compare physical-model-based and traditional synthesis methods by discussing their structural properties. For all structures and models discussed below there are some important general truths. First, a common way of approaching the problem of modeling physical systems is to describe their observed behavior in the frequency domain. Frequency models are particularly effective for the description of linear systems, but such systems rarely apply for musical instruments. When nonlinearities must be taken into account, modeling in the frequency domain often becomes unfeasible, especially when strong nonlinearities are

involved. In this case, models in the time domain are preferable. Moreover, we know that any simulation requires the continuous-time model to be made discrete. This, of course, must be done in such a way as to reproduce with good approximation the behavior of the continuous-time model to which it refers.

Overview of Classical Models

Reasons for Modeling

For us, the aim of synthesis by physical models is to realize models for the generation of sounds, which can be accessed in a natural and intuitive fashion by the user during both the composition process and the performance. In fact, what stimulate interest in this synthesis technique are two fundamental hypotheses: (1) timbral complexity is determined by the model structure and, as a consequence, by the structure of the algorithm that implements the model; and (2) there exists a precise relationship between the reaction of a physical instrument to a certain action, and the reaction of its model. These hypotheses are related to the fact that synthesis by physical models is based on the simulation of the sound production mechanism rather than the sound itself. The first hypothesis suggests that a good physical model should generate a certain timbral complexity like a traditional musical instrument, while the second suggests that the parametric control is easier and more intuitive with an algorithm based on a physical model than with other models (e.g., additive or subtractive synthesis).

The design of an instrument model can be approached in several ways, which, in general, are based on two different viewpoints. One is that of the physicist, who is interested primarily in the study of the sound generation mechanism; therefore this point of view is mainly analytic. The other is that of

the instrument designer, who is interested mainly in the quality of the produced sonorities. This second viewpoint is much more synthesis-oriented, and the implementation of the model results is even more important than the model itself. In fact, a synthetic approach requires the instrument designer to pay particular attention to structures and algorithms.

Physical, Functional, and Formal Structures

To develop a model of a complex system such as a traditional musical instrument, it is helpful to partition the model into simpler submodel "blocks" in such a way that the description can be given in terms of the individual blocks and the modality of their interconnection. There are several ways to accomplish such a subdivision, according to the goal that we want to reach. Among the various possibilities we cite three criteria: (a) physical resemblance, (b) functional structure, and (c) simplicity of the formal description. Partitioning according to physical resemblance consists of the subdivision of the instrument in parts that are physically simple to describe. For example, the violin can be subdivided in strings, bridge, sound board, and bow. In the case of partitioning according to functional structure, the musical instrument is divided into two parts that assume the function of the "excitation," the part of the instrument that causes and sometimes supports the vibratory phenomenon, and that of the "resonator," the site where the musically interesting vibratory phenomena take place. An example of such a functional separation can be given for a violin in terms of the bow, which represents the excitation, and of the combination of sound board and strings, which constitutes the resonator. When partitioning according to the simplicity of the formal description, the model is also divided into blocks, but the subdivision is according to the formal characteristics of the model and, in particular, according to the simplicity of the resulting equations. This subdivision thus always occurs a posteriori, that is, once the model equations are derived. For example, we might divide a model in linear and nonlinear parts.

All of these criteria share a common functional interpretation of the elements of the model decomposi-

tion. For example, a violin may be easily and naturally decomposed as in (a), but if we identify strings, bridge, and sound board with the linear part of the instrument, and the bow with the nonlinear part, we get a decomposition of the kind described in (c). Finally, if we identify the linear parts of the instrument with the resonator and the nonlinear parts with the excitation, we obtain a functional decomposition, like the one described in (b). Even though the excitation and the resonator that arise from the application of this procedure are intended in a generalized sense, the choice of identifying the resonator with the linear part and the excitation with the nonlinear part is not coincidental. In fact, applying this procedure to any traditional instrument we always get a functional decomposition that agrees with our intuition.

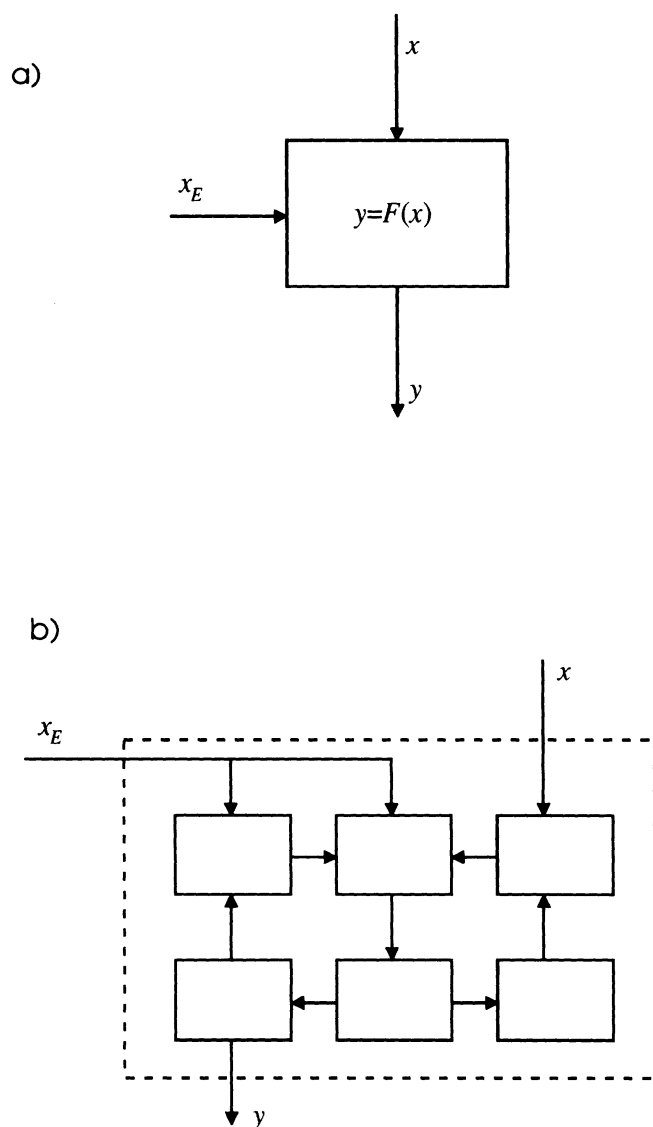
White-Box and Black-Box Approaches

Developing a model for physical synthesis means solving essentially two problems: find a suitable description of the blocks and specify the modality of their interconnection. We will first approach the problem of describing the blocks independently from each other, and then we will discuss how to properly couple them. There are two extreme strategies for the description of the individual blocks: black boxes and white boxes.

Black Box

In this case the model is described only by an input/output relationship, as shown in Fig. 1a. This drastically limits the choice of the signals that can be involved in the description of the model. For instance, we can think of a perturbed string in terms of local displacement, transverse velocity, transverse force, and so on, while, with this approach, the signals must be chosen a priori according to the available I/O relationship. Furthermore, a black-box strategy makes it difficult to choose properly the operative conditions of the model. For example, it would be very difficult to choose freely the position from which a distributed structure like a reed or a string is observed. In general, aspects like those mentioned

Fig. 1. (a) Black-box scheme; (b) white-box scheme.



above may dramatically affect the behavior of the model. Therefore a black-box strategy often results in a model that is too rigid to be useful for efficient synthesis.

White Box

What is described in the white-box case is the whole mechanism of generation, as depicted in Fig. 1b. As a consequence, this synthesis strategy results in mod-

els that are flexible, but difficult to apply. For example, we may describe a string by means of the differential equations that govern its motion. What we get is a set of equations that accurately describe the system but are very difficult to solve for most realistic cases. With this strategy, the number of elements to be simulated often grows quite rapidly as we try to refine the model because it is necessary for the structure of the mechanism to be spatially quantized. On the one hand, this category of models allows the user to access its structure in all of its parts, but it is necessary, on the other hand, to simulate the motion in all of its parts. This is most often an unnecessary requirement that results in an unmotivated increase in the complexity of the model and of its implementation. In both cases, the problem reduces to that of the identification and synthesis of a dynamical system (linear or nonlinear).

Interconnection

The various parts of a model are connected to each other and to the "outside world" (in our case). These interconnections carry information that is unidirectional (feed-forward) in the simplest case. Physically this means that no reaction corresponds to the action exerted by one block on another. When the reaction cannot be neglected, we have a bidirectional exchange of information, which corresponds to a feedback interconnection. In this case the dynamic behavior is generally more varied and difficult to forecast. If there are more than two blocks, they can be composed using either unidirectional or bidirectional interconnections. Notice, however, that even if all connections are feed-forward, the resulting whole structure can be feedback.

Physical instruments must interact with the outside world, as they, on the one hand, must be controlled by the performer and, on the other, must generate a sound to radiate. In the model it is common practice to consider the actions of the performer as feed-forward, even though more sophisticated feedback models have been proposed by Cadoz, Luciani, and Floreas (1984). Moreover, in a real instrument, the sound is diffused because of the vibra-

tion of some surface in air, while in simulated models it is fairly common to listen to the sound which is taken from a precise point of the model and diffused by loudspeakers. Adrien (1990) is currently working on the problem of how to improve this traditionally accepted way of making synthesized sounds listenable.

Excitation

There are several criteria to use in finding a suitable block decomposition of a physical model, and all of them are generally isomorphic to a functional decomposition that splits the model into two blocks called *resonator* and *excitation*. This decomposition, besides being very general, is particularly convenient for synthesis purposes, as our intuition can be very helpful in specifying models of the individual parts. We will analyze in detail the various approaches that can be followed for the synthesis of excitations and resonators below, with reference to the functional interpretation of the elements that constitute the model of the musical instrument.

Setting the Initial Conditions

The method of choosing the initial conditions for a model is applicable to resonators whose response manifests itself as a free evolution. As mentioned above a resonator can always be considered linear. To be more precise, a resonator can be treated—without loss of generality—as a linear dynamical system. As such, it cannot robustly allow for more than one equilibrium state, which, in order to be of practical interest, must be a “stable focus”; this means that any trajectory in the state space must be a convergent spiral. In other words, whatever the initial condition may be, the variables that we end up observing will oscillate. The method of setting up the initial conditions consists of making the system begin in a state that is different from the equilibrium, then observing the unforced trajectory of the system. That the equilibrium of the resonator must be a stable focus is not strictly true, as we can also

allow for infinitely oscillating trajectories. In the first case we have a “dissipative” resonator; in the second case we have the less realistic case of an undamped oscillating system.

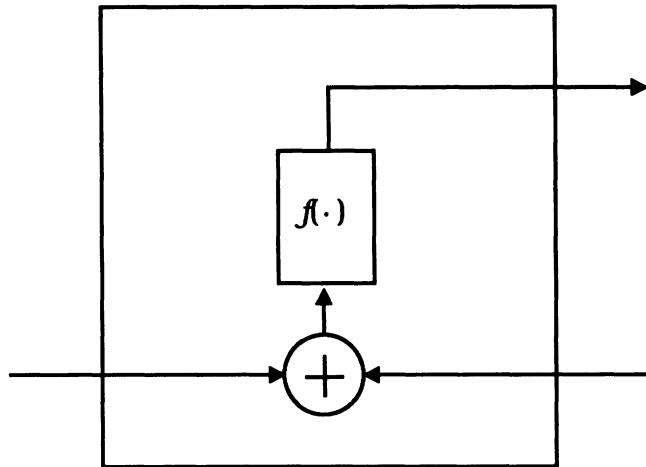
Let us consider the case of the plucked string. We can imagine the string’s having a certain initial distribution of displacements with respect to the rest condition. A realistic initial distribution could be “triangular” (with one edge at the plucking point). The initial state of the system thus corresponds to that distribution of displacements and zero initial velocities of all the points of the string. When the string is released the system state, unforced, begins to evolve (Hiller and Ruiz 1971). A synthesis technique that roughly models this is the Karplus-Strong algorithm, in which the initial condition is set at random by initializing the state variables with random values (“noise”).

A synthesis technique in which the choice of initial condition is important appears to lend itself quite well to those instruments in which the free evolution of states is preponderant, while it is not as useful for persistently excited instruments. Moreover, the transients one gets from such a model are often rather dull from a musical point of view.

Direct Generation

For persistently excited instruments we think of the excitation not just as a “device” that provides the system with a particular initial state, but as a well-defined system acting on a separate resonator. In certain cases we can assume that such an excitation is simply a black box whose behavior is not influenced by that of the resonator. We can use any system able to generate the desired excitation signal, rather than focusing on a model of the excitation. A classic and efficient example is given by table-lookup generators. We can include in the direct generation class those excitations for which a weak feedback from the resonator is allowed in terms of slow variation of the excitation parameters. Notice that the signals that come from gestural actions of the performer on the input devices of the instrument may be treated as excitation signals and generated with a system which can be included in this class.

Fig. 2. An example of a memoryless nonlinear model of an excitation, the reed of a clarinet: the function is defined as $u(t) = A(y + H)^{4/3} (p_M - p)^{2/3}$ when $y > H$, and $u(t) = 0$ when $y < H$.



Memoryless Nonlinear Model

Sometimes it is possible to describe the excitation by means of a relationship such as $y(t) = f[x(t), x_E(t)]$, where $y(t)$ is the "excitation signal" and $x(t)$ is the corresponding "response" of the resonator. In this expression, $x_E(t)$ represents an "external" input signal that normally incorporates the excitation actions of the performer, and $f()$ is a nonlinear function. Since such a function is memoryless, this model neglects any dynamic behavior of the excitation element (McIntyre, Schumacher, and Woodhouse 1983). For the clarinet, for example, it is possible to describe the behavior of the reed by means of an instantaneous function $f(p, p_M)$, which determines the air flow entering the acoustic tube in terms of the pressure p_M of the musician's mouth and the pressure p at the entrance of the acoustic tube, as illustrated in Fig. 2.

In general, the shape of the function $f()$ depends on several physical parameters of the excitation. With the clarinet, for example, changing the force exerted on the reed by the performer's lips may result in a dramatic variation of the shape of the curve $f()$. Even if this is not a problem from a theoretical point of view, such parametric dependence does not allow us to implement $f()$ in table-lookup form, which may result in high computational cost for simulating the model.

The principal novelty of this model with respect to the previous ones is that this model makes use of the

information coming from the resonator, which describes its reaction to the excitation. Another new aspect of this technique is given by the presence of nonlinearity, which, even if memoryless, gives rise to particularly interesting behavior, especially during the initial transient (attack) (Smith 1986).

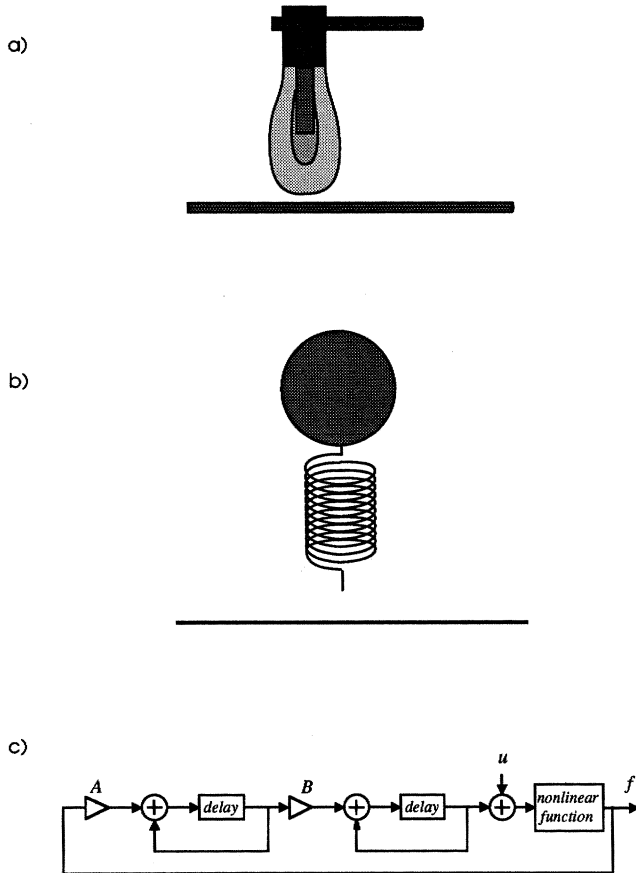
Mechanical Model

Another way to think of excitation is as a component described by a network of ideal mechanical units such as springs, masses, and friction elements. Roughly speaking, the masses are used to model the inertial behavior of the excitation. The springs (which generally have a nonlinear characteristic function) take into account the elastic properties of the body of the excitation, and the friction elements incorporate its internal losses. The excitation is thus described by a set of differential equations that govern the dynamic behavior of these elements. Notice that this model is not memoryless and therefore requires the specification of initial conditions. This may result in a better approximation of the physical system we are modeling.

A simple mechanical model of an excitation is given by a mass and a nonlinear spring. Such a model can be used, for example, to describe the hammer of a piano (Suzuki 1987), and lends itself quite well for modeling excitations that interact with a resonator (Borin et al. 1989), as shown in Fig. 3. In fact, it provides a "force" signal to the string that depends on the relative position of the string itself. This model can also take into account external parameters and/or force signals that determine its motion. Notice that a further nonlinearity is given by the "contact condition" between excitation and resonator. This nonlinearity is generally quite strong as it almost always includes a "step function" derived from the mechanical properties of the underlying system.

The class of mechanical models of excitation is rather general; in fact, a number of models of the bow, the reed, the lips, the tongue (Adrien 1988), and even of the glottis are available in literature. Each of these models can be improved by adding complexity to the system. A very high level of detail can be

Fig. 3. An example of a mechanical model of an excitation, the hammer of a piano: (a) physical hammer, (b) model, (c) algorithm.



reached by means of a regular spatial subdivision of the physical structure of the excitation in massive elements, which are connected to each other by means of springs and friction elements. This method, however, which was successfully implemented in real time by Cadoz, requires (at present) specialized hardware architectures because of its very high computational cost.

Resonators

The description of a resonator, without serious loss of generality, is reducible to that of a causal, linear, and time-varying dynamical system. The linear behavior, which is verified with excellent approximation in all cases of interest, is a strong hypothesis upon which many useful assumptions for analysis

and synthesis can be based. Among the numerous models for the analysis of linear systems are some approaches that lend themselves particularly well to musical synthesis. A short illustration of these methods is presented below.

Transfer Function Models

A "black-box" approach to the simulation of a resonator makes use of its functional input/output relationship, which, for the linearity assumption, can be expressed in terms of transfer function. Clearly, as a result of the black-box approach, the physical structure of the resonator is completely ignored. The resonator is thus a generic device that elaborates an input signal provided by the excitation (Schumacher 1981). For the sake of concreteness, we can think of the black-box model of a resonator as the implementation of a transformation of pairs of dual variables, such as pressure and flow or velocity and force, which are propagated inside the resonator itself. Such a transformation consists of the convolution of the input signal with a certain kernel function (the impulse response).

A particularly interesting alternative technique is based on the description of the homogeneous quantities that are propagated in the resonator in terms of incident, transmitted, and reflected waveforms. For many kinds of resonators, this description allows us to efficiently implement the transfer function and to identify it in several ways. For example, we can determine the impulse response analytically if we know the equations that govern the physical behavior of the resonator (Nakamura and Iwaoka 1986), or experimentally by measuring the impulse response or the complex acoustic impedance. The principal problem with this technique is that each point of the resonator can be described by a different transfer function. This means that for each point that accesses the resonator structure it will be necessary to define a different filter. Further, we must not forget that, in most cases of interest, the transformation is time varying—even a small variation of the model parameters usually results in a substantial modification of the resonator filter. Clearly a musical instrument must be able to withstand continuous

variations of such parameters as the length of the string in a violin (corresponding to the position of the finger of the performer), or the length of the acoustic tube (corresponding to the keys that are pressed) in a trumpet. These are the variations that carry the musical information!

Mechanical Models

Simulation of the resonator structure based on a mechanical model can be considered as complementary with respect to the method just presented. In fact, it is based on precise hypotheses which are drawn from the mechanical structure of the resonator; a set of differential equations, whose solution represents the signal of interest, is derived from the physical laws that govern the behavior of the reference system. The resonator implementation is the solution of these equations (Hiller and Ruiz 1971). One class of models comprises those that obey the classical equations of vibrating bodies. These are partial differential equations in the spatial coordinates and in time. In some simple cases it is possible to find the solution analytically. In general, however, it is necessary to solve the equations numerically, whereby they must be appropriately quantized.

In addition to these models it is appropriate to mention a similar approach, the CORDIS system (Cadoz, Luciani, and Florens 1984), which can be considered a technique of simulation by mechanical model. This method is based on the decomposition of the excitation into ideal mechanical elements. Even if this method lends itself well to the simulation of several kinds of vibrating bodies such as membranes, strings, bars, and plates, it does not lend itself easily the simulation of acoustic tubes or, in general, wind instruments. This model is able to describe the physical structure of a resonator very accurately, but it incurs a very high computational cost if it is required to describe the motion of "all" points of the resonator. This seems not to be necessary for deriving the musical information we need. In fact, the sound of musical instruments, in general, can be related to the motion of a few important points of the resonator.

Waveguide Models

The waveguide model is particularly efficient for modeling resonators; it is based on the analytic solution of the equation that describes the propagation of waves in a medium (Smith 1986, 1992). This model is realized with delay lines, junction elements, and filters and, as such, corresponds closely to our perception of physical reality. In particular, by using these elements, complex structures can be built, such as the bore of a clarinet, with holes and bell, or groups of strings that are coupled on a "resistive" bridge (Garnett 1987).

Modal Synthesis Models

Another kind of resonator can be obtained by means of "modal synthesis" (Adrien 1991). In this kind of model, techniques derived from system theory are used to reduce a linear system to a collection of parallels of second-order systems, each of which is realized as a damped oscillator. Thus a certain modularity and structural regularity are maintained. This technique is "musician friendly" as it has control parameters reminiscent of the classic methods of additive and subtractive synthesis. In fact, the parallel modes can be thought of as sounds coming from resonant filters. The main drawbacks of this method are the difficulty of handling the modal parameters and the complexity of the description of modal structure.

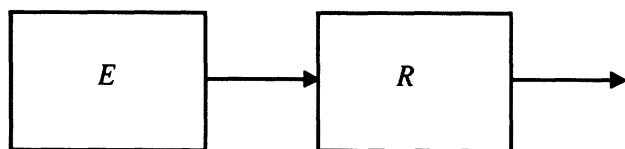
Interaction

Having discussed several strategies for modeling the two functional blocks of musical instruments (excitation and resonator), we now consider the problem of coupling these parts.

Feed-forward

The simplest coupling structure for the functional blocks of a synthesis model is the feed-forward scheme, where the action of the excitation on the

Fig. 4. Feed-forward scheme.

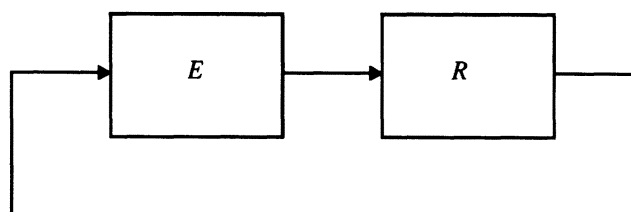


resonator is a unidirectional transfer of information as shown in Fig. 4. This structure lends itself particularly well for describing those interaction mechanisms in which the excitation imposes an initial condition on the resonator and then leaves it free to evolve on its own, or in which the excitation can be treated as a signal generator. In this last case we get a “source plus linear filter” structure typical of feed-forward methods. An example is given by any model of human speech synthesis in which the model of the excitation, represented by the glottis, acts on a model of oral and nasal cavities (resonator) according to a feed-forward scheme. Usually the resonator is a simple filter, which makes the scheme a feed-forward subtractive synthesis model. In this situation, the excitation is controlled by the performer and does not receive any information from the resonator. As a consequence, this model is unable to adequately simulate complex transient behavior.

Feedback

A further development of the feed-forward structure is the feedback interconnection scheme shown in Fig. 5 (McIntyre, Schumacher, and Woodhouse 1983), where the excitation takes into account the state of the resonator, affording a full exchange of information between the two blocks. Most traditional instruments behave according to a scheme of this kind. An example in which the mutual dependence between the excitation and the resonator signals is evident is given by the clarinet. In this instrument the site of the vibratory phenomena is the bore, where the perturbations are due to the variations of the entering flow (action). This flow, however, depends on the aperture of the reed, which, in turn, is a function of the

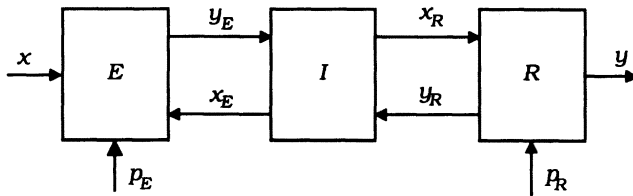
Fig. 5. Feedback scheme.



difference of pressure between the mouth of the performer and the entrance of the acoustic tube (reaction). Therefore, the entering air flow and the mouthpiece pressure are mutually dependent. This example shows that the feedback scheme lends itself well to modeling persistently excited acoustic instruments. Note that, even when free evolution seems to be predominant, as in the piano sound, the interaction between excitation and resonator is critical in modeling transients where, from the point of view of perception, it is crucial to the timbre of the instrument.

The feedback scheme is able to describe the behavior of an instrument very accurately because the interaction between excitation and resonator is the principal element responsible for the timbral dynamic evolution; however, it also has several drawbacks. In particular, because of its generality, its application is sometimes difficult. In fact, to describe the blocks and the mode of their interconnection is not a straightforward procedure. Realizing a feedback scheme may result in a situation in which, in order to preserve the descriptive modularity, even if we are combining explicit equations, we get model equations that are in implicit form and cannot be elaborated. This happens every time the output of a block that provides the feedback information to another block is instantaneously dependent on the input that receives information from that other block. In fact, in this case, there is a closed loop without delay, which gives rise to problems of noncomputability. Another problem that arises when analyzing the structural properties of the feedback scheme is how to guarantee real compatibility between the systems to be interconnected. In fact, normally, the feedback scheme does not allow blocks to be built independently of each other.

Fig. 6. Interaction: three-block scheme.



Modular Interaction

We have pointed out the major problems inherent in the feedback structure. These problems can be dealt with by a synthesis structure based on a three-block scheme (Borin, De Poli, and Sarti 1992; Borin et al. 1989, 1992) as illustrated in Fig. 6. We assume that the E and R blocks, which stand for excitation and resonator, respectively, are both time-varying discrete dynamical systems; R is normally linear, while no restrictions are made on E . Note that, as a pair of discrete dynamic systems, E and R are described by means of explicit relationships. The vector x is the external input of the excitation and p_E and p_R are vectors of parameters that, in general, are time-varying. Roughly speaking, it appears natural to use the external input x to describe the actions that the musician performs to produce the sound, while the parameters p_E and p_R can be used to describe the actions that *modulate* the sound. The output vector y represents the musical signals, while the other vectors carry the exchange of information between E and R through I . The interconnection block I acts as an interface; its main purpose is to separate the excitation from the resonator, so that they can be designed independently. To accomplish this, block I governs the information exchange to ensure that the output of E is compatible with the input of R , and vice versa.

Notice that this scheme preserves the feedback structure; therefore it has the same description capability as the feedback scheme described above. As we can see, there is a certain degree of modularity due to the interconnection block I , whose task is to adapt the exchanged signals with various conditioning operations, such as scaling, delay, integration, or differentiation. In other words, it behaves like a decoupling element, mediating the relationship of the excita-

tion and the resonator. Block I also offers an approximate solution, acceptable for sufficiently high sampling rates, to the problem of noncomputability due to the topology of interconnection. It consists of inserting delay elements into the delay-free loops; this solution preserves the modularity without affecting the computational efficiency. As the excitation and the resonator are dynamical systems, the structure guarantees that their definitions are given in an explicit form; therefore possible computability problems are clustered into block I .

After implementing several models according to the above structure, we have found that in general the interconnection block I causes a very modest increase of the overall computational cost, as it requires only a few simple operations. Modularity requirements are, however, in conflict with the desire to undertake a global optimization, which may reflect on the model's computational efficiency. On the other hand, thanks to the decoupling and independence properties, block I allows us to implement E and R in parallel or pipeline hardware architectures. This has a great effect on the overall performance with an improvement of computational efficiency. In general, block I cannot, connect blocks that are arbitrarily chosen. This mechanism of interconnection is proposed in order to make reasonable connections between compatible blocks in the simplest and most direct way.

Physical Models and Synthesis Algorithms

The classical starting point of synthesis by physical models is simulating as accurately as possible the behavior of traditional instruments. In this context synthesizing a sound means choosing a model of a traditional instrument, realizing a good implementation of the methods presented above, and learning how to control it in order to both obtain traditional sounds and possibly to explore some unconventional timbre space using the model. This approach requires the development of very accurate computational models. Moreover, it requires that the musician acquire a certain performing capability by experimentation and interpretation of the parameters

of the model; this is something that a musician does with traditional musical instruments as well. A more experimental method consists of utilizing various modular models to assemble a model of a complex vibrating structure (Calvet, Laurens, and Adrien 1990). In an extreme case it is possible to use modules that are “really” elementary, like combinations of masses, springs, and dampers (Florens and Cadoz 1991), or waveguide elements (Garnett and Mont-Reynaud 1988)

With this approach the physical reality tends to be the reference element for validating the results. The classical synthesis by physical models, besides having a precise scientific meaning, is especially interesting for musical purposes, and there are other (virtually infinite) musical applications of this way of synthesizing sounds; this fact will be clearer after we have examined synthesis by physical models from the sound synthesis viewpoint.

Algorithms as Sound Abstractions

In traditional music, the function of the instrument is twofold; in fact, besides representing the sound generation process, an instrument can be seen as the abstraction of a class of sounds characterized by a timbre, a dynamic behavior, and certain expressive possibilities. We believe that the twofold function of the generation process holds true in general, and can be applied to synthetic instrument models as well (Borin, De Poli, and Sarti 1990a). As far as the sound generation process is concerned, a musical instrument consists of vibrating parts that produce sound and other parts that control it. In the case of synthetic instruments, there is some hardware-software interface that connects the user and the algorithm. The algorithm and its implementation are thus hidden by this interface, which ends up specializing the algorithm itself toward the generation of a certain category of sounds. Ignoring the mechanism that produces the sound, a performer learns how to obtain the desired results by acting on the control mechanisms. The musician thus works only with the interface and—considering the interpretation of the underlying mechanisms provided by the instrument

designer—builds his or her conceptual model of the instrument, and develops a certain performance practice for obtaining the desired sounds.

To understand the function of the model as an abstraction of a class of sounds, we should not forget that it is common sense to classify a timbre or a whole class of timbres according to the mechanism of sound generation. For example, we normally speak of woodwinds meaning those instruments (made originally of wood) in which the mechanism of generation is given by the interaction of an acoustic tube and a “virtual” reed, which is intended as a device whose aim is to modify the air flow according to a difference of pressure (e.g., clarinet and flute). Another example is given by brass instruments, which are characterized by the fact that the excitation is provided by the vibration of the lips of the performer. In a similar way we identify classes of synthetic timbres with the algorithms that generate them. More precisely, while acoustic instruments are characterized by the mechanism of production of the sound, synthetic classes of sounds are characterized by the structure of the algorithm.

The Structure of Algorithms

A good sound is characterized by its complex dynamic behavior, which depends both on external control and on the internal structure of the synthesis scheme. We think that sound complexity which is due mainly to the sound production mechanism is always preferable with respect to a technique in which the desired sonority is obtained by utilizing external control, as it makes the choice of parameters much easier and more natural. If we examine various synthesis algorithms from a structural point of view (De Poli 1992), we find that the simplest structure is that of direct generation. This structure describes all of the techniques that are based on the combination of the outputs of one or more independently operating blocks, as in additive or granular synthesis. Practically speaking, when using algorithms that have this structure, it is necessary to specify many parameters, and the result depends on the coherence with which these parameters are cho-

sen. This coherence is not a built-in property of the structure; it must be guaranteed during the specification of the parameters.

A second category of algorithms is characterized by a feed-forward multiblock structure in which some blocks generate a signal that is supplied to other blocks for a post-processing. This class includes all linear and nonlinear techniques such as subtractive synthesis, amplitude and frequency modulation synthesis, and some early examples of physical model synthesis. An important characteristic of the feed-forward algorithms is a certain inherent sound complexity arising from their structure. In other words, by choosing a synthesis technique in this class, we begin to give the structure the task of producing "complexity" in the synthesized sound. The dynamic behavior, however, must still be specified by the user.

A third class of algorithms is characterized by an interacting multiblock structure. The simplest example of this scheme consists of a pair of blocks coupled in feedback. Synthesis by physical models is a special case of this structure that has a precise physical interpretation as well. This interpretation is useful for identifying the parameters of the controls of the models, and sometimes also for evaluating properties such as stability or convergence. But again, its main advantage is the fact that the properties of the produced sonorities can be identified with the structure of the class of algorithms to which it belongs.

Pseudophysical Models

Physical model synthesis refers to the real world not only to find inspiration for building conceptual models, but also to identify system implementation parameters and, most importantly, to evaluate the results. This is not a peculiar characteristic of this synthesis method; results are often compared with natural sounds for qualitative judgment during synthesis using any technique. But synthesis by physical models has the unique feature of taking this reference as its validating hypothesis. We think this com-

parison limits the possibilities of this technique because, in general, a serious comparison between reality and simulated results cannot give satisfactory results; even a state-of-the-art model is only a rough approximation of a traditional musical instrument. Furthermore, in the system implementation, the computational costs may result in further simplifications to the models that can effectively be simulated. Indeed, references to the real world are somewhat arbitrary and unmotivated, since we give importance mainly to the musical reality (Borin, De Poli, and Sarti 1990b). From our point of view, synthesis by physical models—being a sort of a musical reality "generator" on its own—makes it possible to take inspiration from the real world in order to derive our interpretation of it without forcing us to limit experimentation to the usual physical equations. This leads us to focus our attention on the structures, and marginally on the physical interpretation. Structures are responsible for the generation of homogeneous sounds, while interpretations can be considered as useful or limiting tools, depending on particular situations.

A typical situation in which physical interpretation is important is when we need to specify the control parameters of an instrument. Algorithms that are derived from physical models are inherently robust with respect to control configurations. This property arises from the fact that physical models possess characteristics of passivity and stability, which allow us to make an a priori choice as to the parametric variations allowed and to forecast the effects on the output. Therefore the physical interpretation of the structure, besides facilitating the parametric control of the instrument, provides the musician with a conceptual model that is very close to his or her experience. Starting from these considerations, it becomes extremely interesting to experiment with structures that are not anchored in physical reality and whose only constraints are stability and passivity. These models take physical reality only as a source of inspiration but cannot be strictly considered as physical. For all of these reasons, pseudophysical models represent a field of sound synthesis that has yet to be explored.

Conclusion

Synthesis by physical models, begun as a method for studying the physical behavior of traditional musical instruments, is gaining interest as a method for producing music. There are still many tasks remaining as far as the research of efficient and effective algorithms for simulating the various vibrating structures are concerned. This is especially true when considering the problem of making the algorithms suitable for parameter control purposes and making them useful as aids to our intuition in the choice of control parameters for performance. This can be achieved by providing an easier interpretation of the relationships among parameters, and a physical interpretation lends itself well to this purpose.

Synthesis by physical models can itself be considered as a musical "reality generator." The model should not be interpreted in order to compare the quality of such musical reality with that of traditional instruments, but to help to "conceptualize" such a reality. As a consequence it might be particularly interesting to create structures that are not required to have any physical interpretation and have the only constraint of being stable and passive. In this case physical reality is taken only as a source of inspiration and is not used as a reference for a qualitative judgment of the sonority produced.

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