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# A Virtually Real Flute

Since the first keyboard-controlled digital synthesizers became available, several new synthesis interfaces have been developed (e.g., Mathews 1991a, 1991b; Cook 1992; De Laubier 1998). As most of these digital instruments differ considerably from traditional instruments, musicians must learn new techniques to play them (Kronland-Martinet, Voinier, and Guillemain 1997). Here, we propose overcoming this difficulty by designing a digital flute using a traditional instrument form factor to control a synthesis model. The digital flute was assumed to extend the technical scope of the traditional flute, but we also wanted to be able to use the instrument in the traditional way. To connect the instrument to a computer, we added sensors to its key pads and placed a microphone inside the mouthpiece. The synthesis model to be controlled by this interface had to take the physical characteristics of the instrument into account. A physical model was therefore developed to simulate the propagation of waves inside the flute.

The system of excitation involved in flute-playing is highly complex from a physical point of view. To construct a real-time model with parameters that can be measured while the instrument is being played, we used a signal model to simulate the source excitation. By injecting this model into the physical one, we constructed a hybrid model which accounts for both the physical and perceptual aspects of the sound produced.

## Design of the Interface

Playing a wind instrument involves two main factors. The first of these is the player's finger position, which is correlated with an equivalent length of the instrument (Nederveen 1998) and thus with the pitch, and the way the instrument is excited by the air jet (Fletcher and Rossing 1990; Verge 1995). This information must be detected and combined in order to control a synthesis model in real time or to produce standard MIDI messages.

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### Pitch Detection

The pitch is determined by both the player's finger position and the way the instrument is blown. Signal processing methods can be used to analyze the sound emitted by the instrument and accurately estimate the pitch. Since the flute is a monophonic instrument, a pitch extractor of this kind can be used to directly perform the MIDI encoding of musical sounds without having to solve the problems associated with polyphonic instruments. The `fiddle~` MSP object (Puckette, Apel, and Zicarelli 1998) is a good example of an available tool which is well suited to applications of this kind. In this case, the instrument only needs to be equipped with a microphone connected to the sound input of a computer running an MSP program.

In our case, we wanted to be able to control the synthesis model with the real instrument, even when the flute is not blown. The state of the key pads therefore must be detected to obtain information about the player's finger position. In addition, the key pad noise is of musical relevance, and we therefore had to collect information of another type: the speed at which the key is pressed. To detect the time-varying position of the keys, we used a combination of magnets and Hall effect sensors. A Hall effect sensor gives an output voltage which is a function of the magnetic field received. If the magnetic field is generated by a permanent magnet, its intensity will depend on the square of the distance between the sensor and the magnet. The output voltage of the sensors is then correlated with the spatial distance between the key pads and

Figure 1. Close-up view of the flute equipped with magnets and sensors.

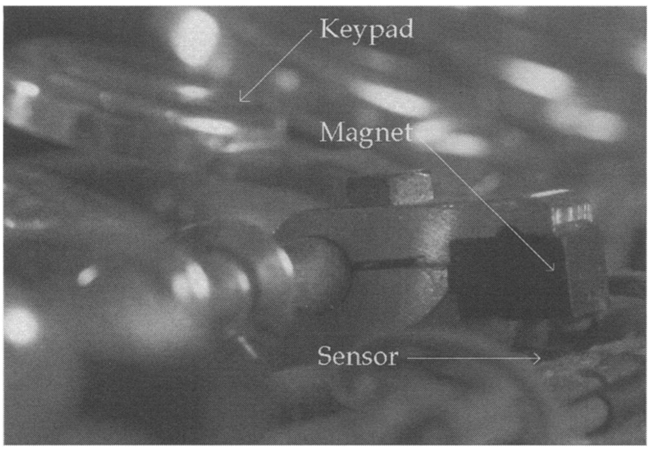
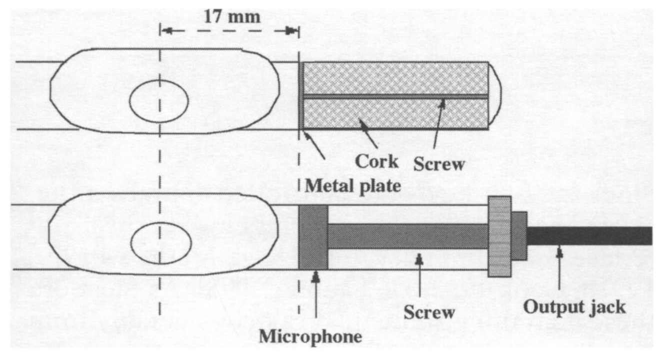



Figure 2. Modification of the mouthpiece of the traditional instrument.




the corresponding holes, which makes it possible to detect whether the hole is opened or closed. By regularly sampling this voltage, we can estimate the speed at which each key is pressed or released. For practical reasons, the magnets were connected to each finger key on the instrument, while the Hall effect sensors were placed in front of each magnet on an aluminum rail placed parallel to the instrument, as shown in Figure 1.

The magnetic field generated by the magnets had to be strong enough to obtain a suitable output signal from the sensors. The two main states of the holes (opened and closed) had to be clearly distinguishable. The magnets were chosen so that the neighboring sensors were not influenced by the magnetic field.

An Infusion Systems I-Cube System was used to digitize data from the sensors. This system digitizes the sensors' output voltages and sends this data to a Macintosh computer running Max. With this interface, it is possible to sample 14 sensors at a rate of 50 Hz with a resolution of 7 bits, which suffices for this application. (A Max object, iCube, is provided with the hardware, and a Max patch can easily be created in order to process the data from the sensors.) The processing consists mainly in checking whether each aperture is in the open or closed state, and then finding the corresponding pitch in a lookup table. In this case, wrong fingering would not be recognized, and the last valid pitch detected would, for example, remain activated.

### Measurement of the Excitation

The way the instrument is excited is a highly complex process. It depends on several parameters, such as the player's lip position and the angle at which the air jet hits the labium of the embouchure. These important features are difficult to measure, thus we decided to concentrate on detecting the internal pressure, which depends on the way the instrument is being played.

To measure the acoustic pressure produced by the instrument, a microphone was placed at the embouchure level (near the mouthpiece). More specifically, the original cork with which the instrument is fitted was removed and replaced by a custom assembly containing the microphone, enabling the instrument to be finely tuned. This device is shown in Figure 2.

A moisture-resistant electrodynamic microphone able to handle high acoustic pressure (approximately 140 dB SPL) was placed inside the flute pipe, and the microphone signal was delivered to the hardware sound input on the Macintosh. The signal was sampled at the audio rate and processed by a peak detector (McNally 1984) providing the pressure envelope. The pressure envelope was then sampled at a lower rate (50 Hz) and used to trigger note-on and note-off MIDI messages in which the associated pitch is given by the state of the holes. A schematic description of the MIDI generator is given in Figure 3.

MIDI compatibility makes it possible to connect the flute interface to other MIDI instruments and to control them in different ways. A flautist uses

Figure 3. Overview of the flute's MIDI component.

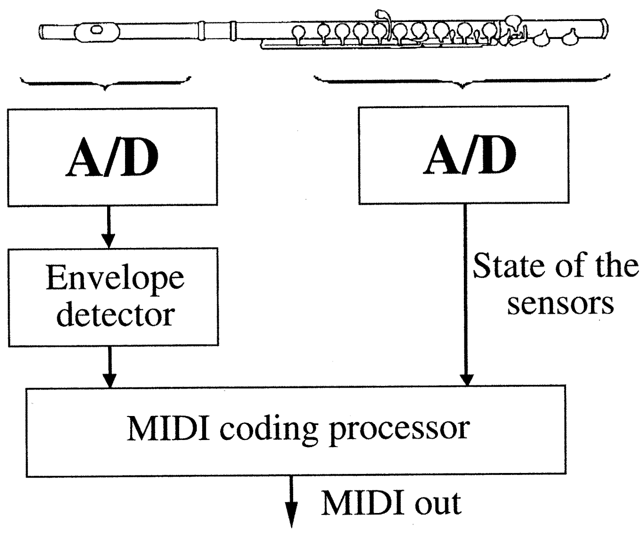
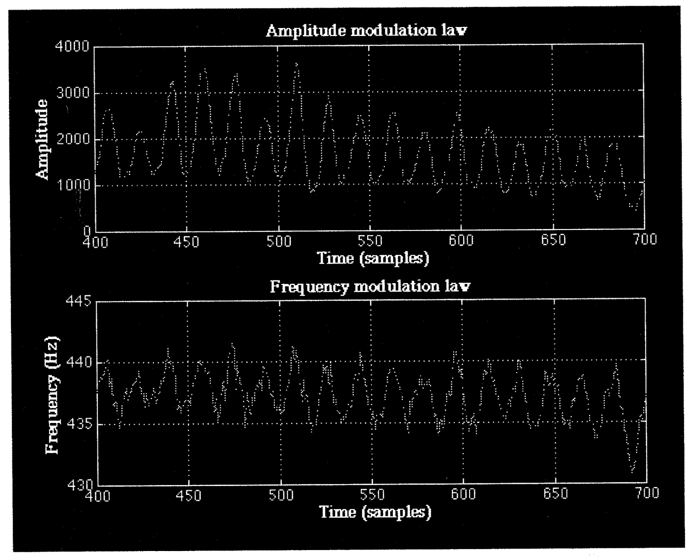



Figure 4. Amplitude and frequency modulation laws of the fundamental component of a flute sound (A4) with vibrato.




pressure variations, for instance, to produce effects such as tremolo and vibrato (Fletcher 1975). To investigate the relationship between pressure variations (tremolo) and frequency variations (vibrato), amplitude and frequency modulation laws can be estimated using time–frequency techniques (Kronland-Martinet and Grossmann 1991). In Figure 4, a close view of the amplitude and frequency modulation laws (fundamental component) of A4 (440 Hz) flute sound with vibrato is shown. We can see that the vibrato is clearly in phase with the amplitude variation (Ystad 1998), which means that the frequency modulation can be controlled by variations in the air jet pressure. With the new instrument, effects of this kind can be used to produce MIDI messages such as aftertouch and pitch bend.

At this stage, the traditional instrument can convert the fingering and even the sounds produced into MIDI information. This method could be used to drive any MIDI synthesizer. It could also be used to study performance and interpretation—or, after some programming—to accompany the flute player with another MIDI instrument. Since the I-Cube system is able to manage 32 inputs, other analog inputs can be used for additional devices that either trigger MIDI messages or control synthesis parameters. Some of these possibilities will be described in the last section of this article.

## The Synthesis Model

Although the MIDI part of this interface is of musical interest, its main advantage is the fact that it can be used to control sound synthesis models for wind instruments in a natural way. These synthesis models must be constructed with the specific interface in mind. This means that the model must be able to operate in real-time and that its parameters must be linked to the interface in a natural way. We decided to implement a source-resonance model in which both physical and signal models are combined (Ystad 1998, 2000). This synthesis model takes into account many physical features of the sound-producing system as well as many perceptual features captured in the spectral representation.

### Physical Model

The propagation of waves inside the flute can be simulated using physical models. These models can either be constructed from the equations describing the behavior of the waves propagating in the structure and their radiation in air (Chaigne 1995) or from the behavior of the solution of the same equations (Karjalainen et al. 1991; Cook 1992; Smith 1992). We opted for the latter alternative by constructing a waveguide model consisting

*Figure 5. The waveguide model.*

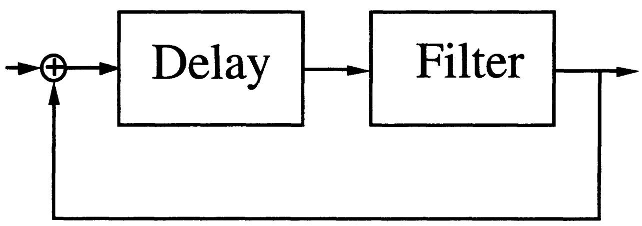
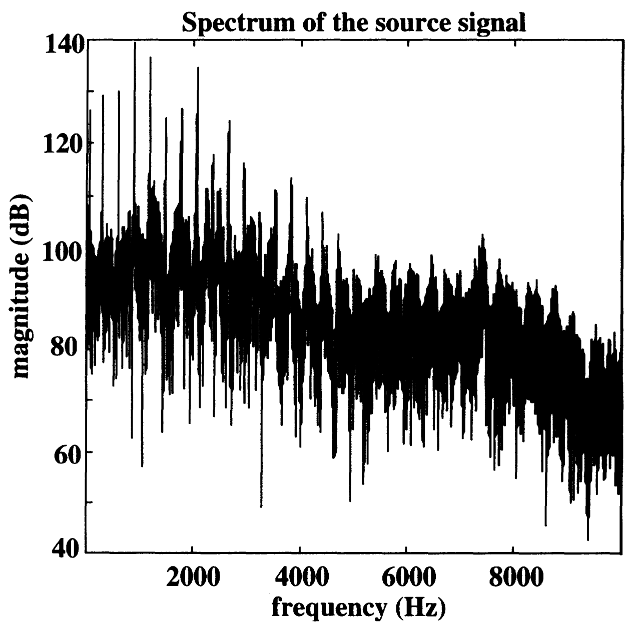



*Figure 6. Spectrum of the source signal of a flute sound (D1 with a dynamic level corresponding to mf) obtained by deconvolution.*




of a looped system with a delay line simulating the propagation time of the waves and a filter simulating both dissipation and dispersion phenomena. The waveguide model is shown in Figure 5.

We have proposed a new way of constructing the loop filter (Ystad 1998) related to the inverse of a time–frequency representation for a transient sound (Guillemain and Kronland-Martinet 1996). To construct the filter, the damping factors and the frequencies of the eigenmodes in the tube must be found. These parameters can either be calculated from theoretical equations describing waves propagating inside a tube (Kergomard 1981; Fletcher and Rossing 1990) or be obtained by performing time–frequency analysis on real sounds (Kronland-Martinet 1988). In a performance situation, the player can control these parameters via sliders. This gives the user the possibility of changing the characteristics of the resonator of the instrument during the performance. One can thus make cross synthesis by combining the flute source excitation with the resonator of any instrument (or conversely), or doing continuous morphing—for instance between a flute-like sound and a string-like sound. The finger position (which is detected by the sensors on the flute) is used to compute the value of the delay line.

The physical processes involved in the excitation of a flute instrument are much more complex. The air jet from the player's mouth hits the labium of the embouchure, and this interaction transfers energy to the acoustic standing waves in the resonator (Coltman 1968; Verge 1995). Flow visualizations of the jet/labium interactions have shown the occurrence of vortical structures on each side of the jet (Fabre, Hirschberg, and Wijmands 1996). This means that a complete physical model of the excitation system would be very difficult to implement in real-time. In addition, the parameters involved in the equations would be difficult to measure while the flute is being played. Therefore we decided to use signal models to simulate the source.

## Signal Model

To construct a signal model simulating the excitation of a flute, we first had to extract the source from the rest of the flute sound. From a physical point of view, the source and the resonator cannot be separated because they interact constantly while the instrument is being played. In our case, however, the separation of the source and the resonator turned out to be a good approximation. As mentioned, we previously developed a model simulating the resonator of the instrument. By removing the contribution of the resonator from the total signal, we obtained the source signal. Because the transfer function of the physical model corresponds to a recursive all-pole filter, we know that its inverse exists. This means that we can extract the source signal from the total flute sound by deconvolution (Ystad 1998). Figure 6 shows the deconvolved signal extracted from a flute sound.

This figure shows that the spectrum of the source signal contains both spectral lines (harmonics) and a broadband noise (which will be called in what follows the *deterministic* and the *stochastic* contributions, respectively). We proposed to separate these two contributions to model them independently. Among the many methods available for this purpose (e.g., Serra 1989), the Least Mean Square (LMS) algorithm, which uses an adaptive filter, was found to be the most suitable for dealing with the problem (Widrow and Stearns 1985). This method involves removing all the components from an input signal that are correlated with a reference signal. By using an estimation of the deterministic signal (harmonics) as a reference signal and the source signal as an input signal, we obtained the stochastic signal (Ystad 1998, 2000).

### *Modeling the Deterministic Part*

The deterministic part of the source signal was found to have nonlinear behavior, because the amplitudes of the spectral components evolve differently from each other as the excitation energy increases. This is the case for most musical instruments, whose timbres depend greatly on the dynamic nature of the sound excitation. In most cases, the nonlinear behavior is correlated with the excitation, and we assume this to be the case here. To model these nonlinearities, we used a global synthesis method, namely the waveshaping method (LeBrun 1979; Arfib 1979), because it provides a useful means to generate complex spectra from easy calculations by performing only a small number of operations. This method consists of distorting a sinusoidal function with an amplitude function $I(t)$ (called the index of distortion) with a nonlinear function $\gamma$. The function $\gamma$ can easily be linked to the spectrum of the sound generated for an index value $I(t) = 1$. In this case, the coefficients of the Chebyshev decomposition of $\gamma$ are given by the values of the modulus of the spectrum to be generated:

$$\gamma(\cos\omega_0 t) = \sum_{k=0}^{\infty} \alpha_k T_k(\cos\omega_0 t) = \sum_{k=0}^{\infty} \alpha_k \cos k\omega_0 t \qquad (1)$$

The index of distortion is said to be bounded ($-1 \leq I(t) \leq 1$), and the waveshaping function will be chosen so that the synthetic signal obtained for $I(t) = 1$ corresponds to the richest part of the real signal (i.e., a *fortissimo* sound). The goal is then to associate with the waveshaping index a measurable value such as the driving pressure to control the spectral evolution of the synthetic signal. One great disadvantage of the global synthesis technique is that the representation of signals is not complete, and it is therefore not possible to reconstruct any spectral evolution by simply changing the index. Nevertheless, as we shall see, the index can be estimated so that the reconstructed signal satisfies perceptual criteria.

One well-known perceptual criterion is the spectral centroid criterion (Beauchamp 1982). It relates to the brightness of a sound and is given by the first order moment of the spectrum. We first applied this criterion to the case of the flute, but discovered that it did not work because very few of the components (mainly the first through sixth) in a flute spectrum change with the dynamic level of the sound. This means that greater importance must be given to these components. We therefore adopted another perceptual criterion called the tristimulus criterion (Pollard and Jansson 1982). This criterion deals with the loudness of three separate parts of the spectrum: one where the evolution of the fundamental is considered; one where the second, third, and fourth components are considered; and one where the rest of the components are considered. The loudness value of each group can be computed using Stevens' formula (Stevens 1972):

$$N_i^n = 0.85 N_{\max} + 0.15 \sum_i^n N_i \qquad (2)$$

where $N_i^n$ is the required equivalent loudness (for the group which contains components $i$ through $n$), $N_{\max}$ is the loudest part of the group, and $\sum_i^n N_i$ is the loudness of all the partials in the group. The total loudness $N$ of the sound is then given by the sum of the three loudness groups:

$$N = N_1 + N_2^4 + N_5^n \qquad (3)$$

With this method, the tristimulus can be given in an acoustic tristimulus diagram, where

$$x = \frac{N_5^n}{N}$$

*Figure 7. Tristimulus diagram of real(+) and synthetic(*) flute sounds.*

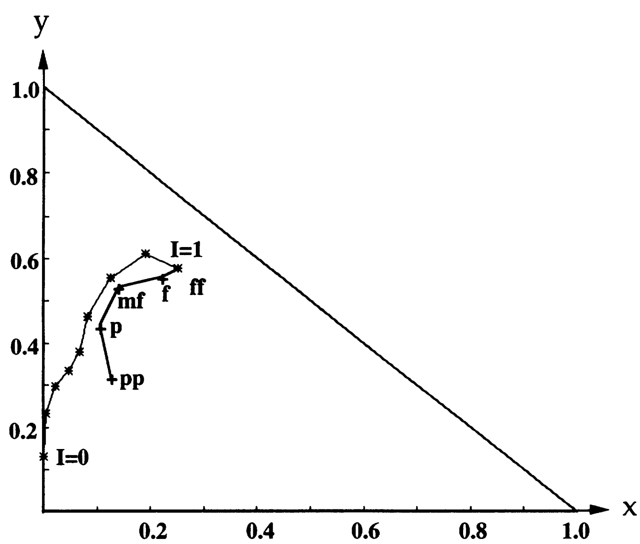
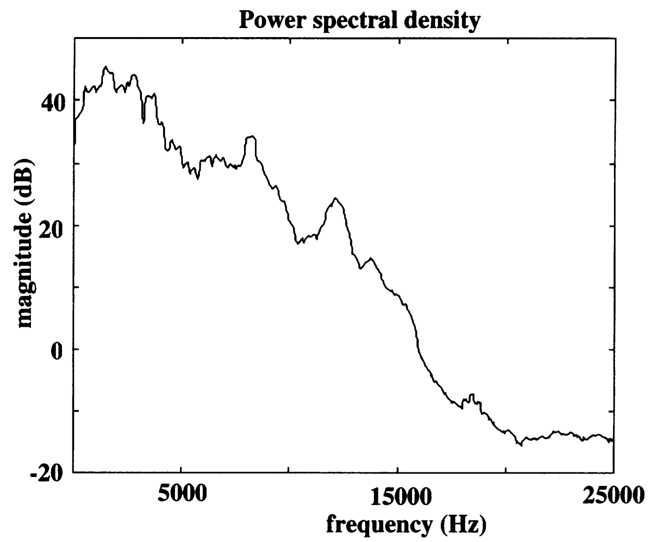



*Figure 8. Power spectral density of the stochastic part of the source signal (the same pitch as in Figure 6).*




$$y = \frac{N_2^4}{N},$$

and

$$z = \frac{N_1}{N}.$$

Since $x + y + z = 1$, it is sufficient to use two of the coordinates($x$ and $y$) to draw the tristimulus diagram as shown in Figure 7. Here, the tristimulus of the sounds generated by waveshaping synthesis with index values ranging from 0 to 1 are represented, along with five flute sounds with different dynamic levels (*pianissimo* to *fortissimo*). The nonlinear function was chosen so that the spectrum generated would coincide with the real fortissimo spectrum for $I = 1$. Since the tristimulus is a perceptual criterion, only its global behavior is important. This means that the real and the synthetic sounds will not show exactly the same behavior, but they will be located in the same area of the diagram and have the same global evolution.

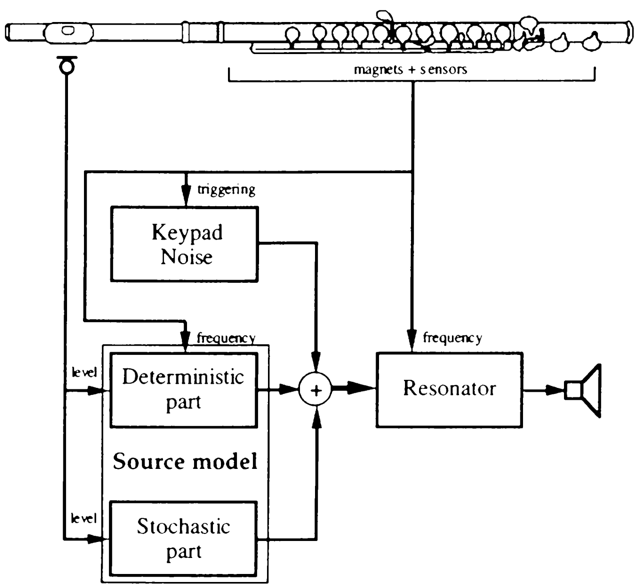
By minimizing the difference between the real and the synthetic flute sounds, we observed that the index of the waveshaping function tends to vary from $I = 0.5$ to $I = 1$, depending on the logarithm of the driving pressure. Consequently, the spectral evolution of the source signal can be controlled by the pressure from the player's mouth, which was measured with the microphone replacing the cork.

### *Modeling the Stochastic Part*

The stochastic part of the source signal was assumed to be stationary and ergodic. That is, we assumed the excitation noise can be described by its power spectral density and its probability density function. From the perceptual point of view, the "coloring" of the noise is mainly related to its power spectral density. Its probability density function $f_B(x)$ can also be relevant. It is related to the histogram of the values $\chi$ involved in the noisy process $B$. It can be easily estimated provided that the random process can be separated from the deterministic one, which is generally true in the case of source signals. In the case of the flute, the histogram is symmetric and follows an exponential law. The power spectral density of a flute sound is shown in Figure 8. By filtering broadband noise (the response of which is given by the extracted power spectral density), one can generate a satisfactory flute source noise.

This model, together with the model of the deterministic part of the source signal, gives a general model of the source signal based on signal modeling procedures. By combining the source model with the physical model simulating the behavior of the

*Figure 9. The hybrid model obtained by combining a signal model with a physical model.*




waves while they are propagating through the medium, very general sound models can be constructed. In the next section, we shall see how this general model can be applied to the case of the flute.

## Hybrid Model

We have called the complete model a *hybrid model*, because it is a combination of two classes of synthesis models: signal models and physical models. This is a very powerful model, as it benefits from advantages of both classes of sound models. Figure 9 shows the flute model, consisting of the source model containing the deterministic and stochastic contributions, and the physical model simulating the resonator of the instrument. To complete the picture, we have added a third part to this synthesis model. In this part, the goal was to generate fixed sounds such as the noise produced by the key pads. This was done using a sampler-like method which consists of reading a previously stored signal designed so that its passage through the resonator gives a realistic key pad impulse. This signal can be obtained, for example, from recordings of the real key pad noise. It would also be of musical interest to use sounds from percussive instruments at this stage.

## The Musical Context

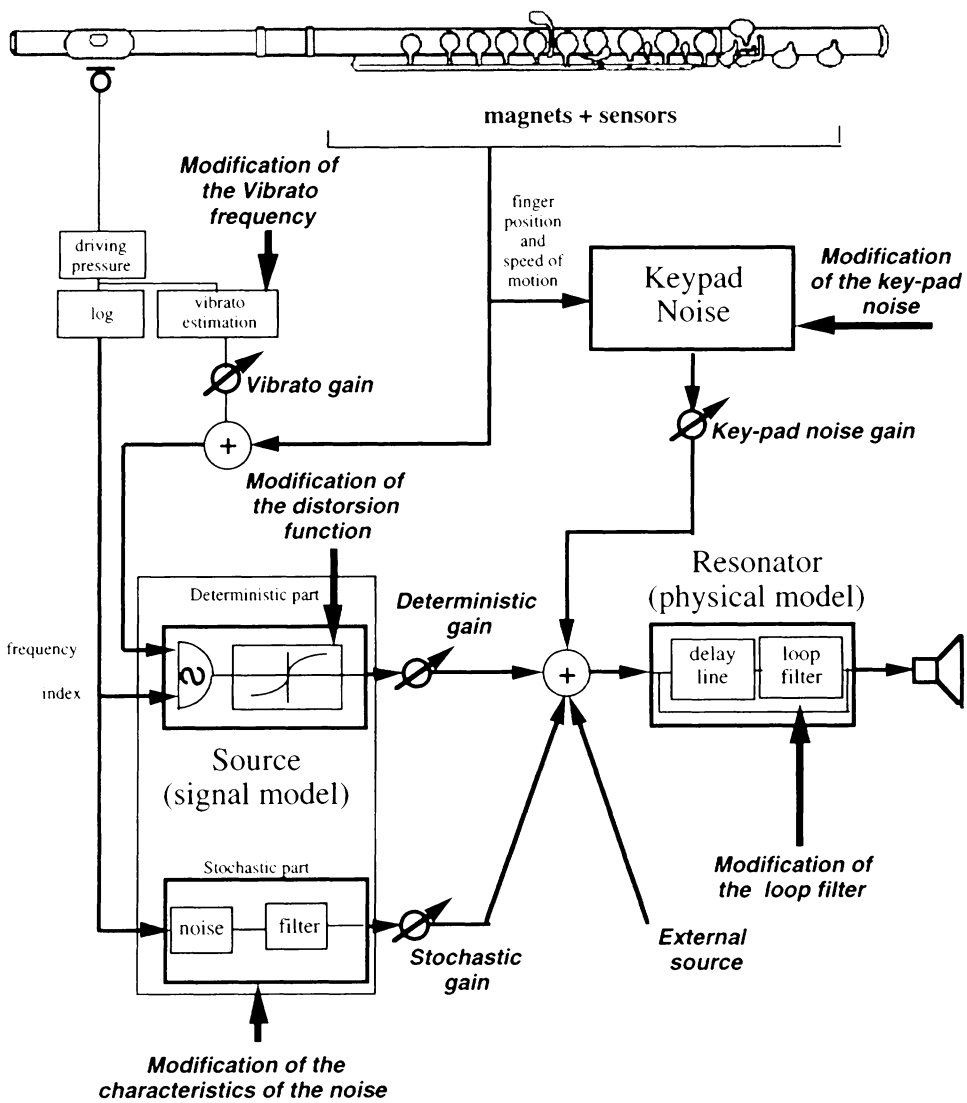
In a musical context, the "virtually real" flute can be played in various ways. It can still be played like a traditional flute, it can be used like a traditional instrument while controlling the synthesis model in real-time, and it can be muted by inserting damping material near the mouthpiece so that only the sounds from the synthesis model can be heard. Additionally, it can simply be used as a MIDI controller to drive other MIDI instruments.

When the instrument is used with the synthesis model, it can effect sound transformations. In the first part of this section, we describe the sound transformations which can be obtained by varying different parameters of the synthesis model. There are many possibilities, and in a performance situation it is therefore necessary to limit the number of parameters to be modified to prevent the instrument from becoming too complicated for the performer. In the second part of the section, we give an example of an application of the instrument where the player is able to modulate four different parameters of the model and where the instrument also controls a Yamaha Disklavier MIDI piano.

As mentioned earlier, the interface was designed so that the possibility of playing the flute using traditional techniques could be retained. This means that musicians can use their hands and mouths to vary the frequency and the pressure, as under normal performance conditions. The player may therefore have to use other parts of the body to regulate the parameters of the synthesis model in order to make sound transformations. The I-Cube System which is used to power the Hall effect sensors connected to the key pads of the instrument makes it possible, for instance, to use volume pedals to control the model's parameters. In Figure 10, we have indicated some of the parameters of the synthesis model which can give interesting sound effects.

The source signal of the flute model comprises three different contributions which must be calculated and mixed before being processed by the resonator. These contributions correspond to the deterministic part, the stochastic part, and the noise generated by the key pads. The deterministic part of the source signal consists of an oscillator

Figure 10. Control of the model's parameters. The possible modifications are shown in an italicized, boldface font.




and a nonlinear function, while the stochastic part consists of filtered noise. The driving pressure controls the stochastic part as well as the amplitude of the oscillator component of the deterministic part. The sensors connected to the key pads detect the player's finger position, and thus controls the delay line of the resonator as well as the frequency of the oscillator. The speed at which the key pads are closed is also detected and used to control the key pad noise. By bandpass filtering the pressure envelope, the vibrato can be estimated and added to the frequency component of the oscillator, causing fluctuations in the resonance peaks of the source. This means that when the source is injected into the resonator, the resonance peaks of the source and those of the resonator will not be tuned all the time. The output amplitude of the system will therefore fluctuate and be stronger when the two systems are tuned than when they are not tuned. The amplitude fluctuations (tremolo) will therefore follow the frequency fluctuations (vibrato) as on a traditional flute.

With all the components that constitute the synthesis model in mind, we can now describe how to produce interesting timbral variations by controlling them independently. By changing the gain of the filter, the depth of the vibrato can be changed. Special shapes of vibrato can also be artificially generated. The waveshaping index can also be controlled. The waveshaping index is a highly sensitive parameter that was estimated to fit the spectral evolution of the flute sound. Nevertheless, a change in the correspondence between the internal pressure and the distortion index can be envisioned. The flute can be given a brassy effect, for example, by increasing the variation domain of the distortion index.

Changing the characteristics of the distortion function dramatically affects the timbre of the deterministic part of the source signal. A distortion function with a decomposition that contains only odd Chebychev polynomials can be used to generate a clarinet-like or pan flute–like source, for example.

The characteristics of the noise can be modified via the noise filter (power spectral density) and the statistics (probability density function). The relationship between the deterministic and stochastic parts of the source signal can also be changed by adjusting the noise gain. If the deterministic part is removed, then the resulting sound would be a noise filtered by the resonator.

The level of the key pad noise can be adjusted by adding a gain to the key pad noise table output. If both the deterministic and stochastic parts of the source are removed, the resulting sound will correspond to that obtained by closing the key pads. The key pad noise can also be altered by modifying the corresponding table and could be replaced by any percussive sound.

One can also use an external input with its own level control to drive the resonator via an external signal. This can be used to take the various noises made by the player's mouth into account, for example.

The loop filter of the resonator can also be adjusted. The loop filter characterizes the resonator and represents dissipation and dispersion phenomena present in the bore. Altering this filter will change the characteristics of the medium in which the waves are propagating. Cross-synthesis effects can be obtained using parameters corresponding to the source of one instrument and the resonator of another instrument. Using a loop filter corresponding to a string with a flute excitation, a very particular sound can be generated, corresponding to blowing "into" a string. Likewise, external sources to be filtered by the resonator can be added. This would make it possible, for example, to generate the noises made by the flautist while playing.

One can also vary the delay line's length and the oscillator's frequency. By changing the offset of these parameters, one can simulate instruments with unrealistic sizes, such as an extremely long flute.

All these manipulations show the advantages of sound modeling. With this particular sound model, one can model not only synthetic sounds, but also natural sounds.

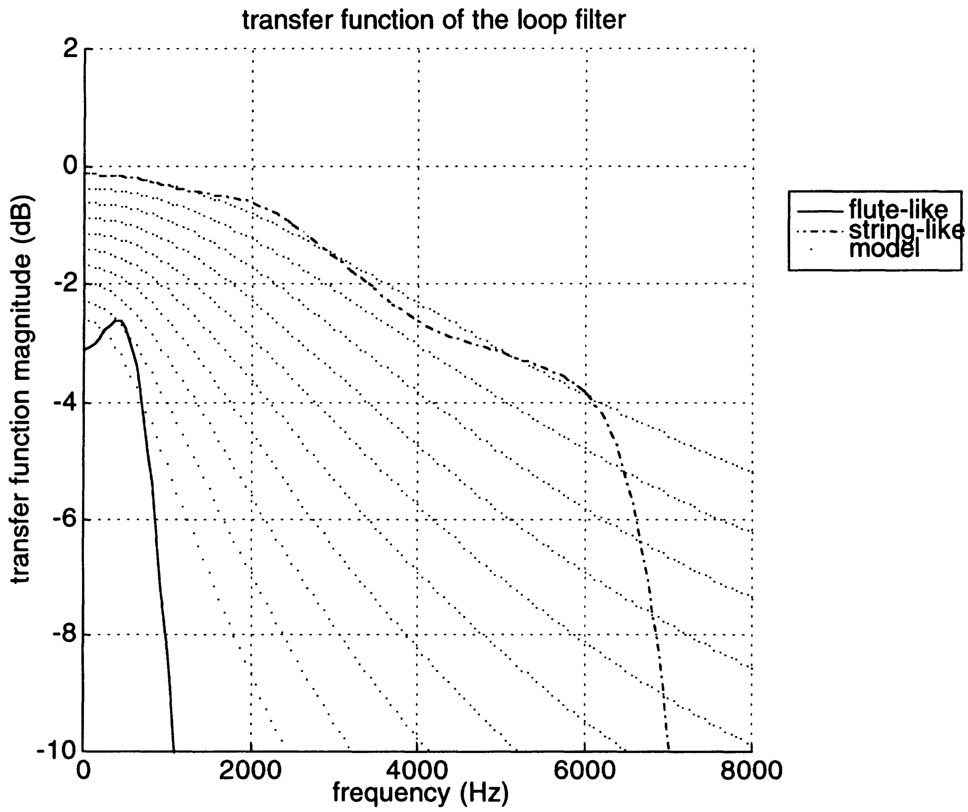
A special thought must be given to the diffusion problem. The instrument has several outputs: one corresponding to the acoustic signal, the others corresponding to the outputs of the synthesis model at different stages. These outputs can be post-processed (with reverberation, equalization, etc.) separately and then diffused or spatialized in the concert hall. This offers many possibilities for live performances, such as independent spatialization effects for each component of the sound. These effects can be controlled either by the flautist (using pedals or specific actions on the instrument), or programmed in advance.

To end this section, we give an example of an application of the digital flute where the player can control four different parameters with pedals, and where the flute also controls a Yamaha Disklavier MIDI piano.

In our setup, the first pedal is used to adjust the ratio between the deterministic and the stochastic parts of the source signal. This makes it possible to make a sound with little or no noise, or to remove the deterministic part in order to obtain a filtered noise.

The second pedal controls the nonlinear function. In fact, the nonlinear function has been split into an odd and an even function, which makes it possible to adjust the amount of odd and even harmonics. A continuous morphing can then be obtained between a pan flute–like sound and a clarinet-like sound. The real flute sound corresponds to a median position.

*Figure 11. Transfer function of the loop filter in the case of a flute-like sound (solid) and a string-like sound (dashed). The dotted curves show various transfer functions of the loop filter obtained by jointly adjusting parameters G and a.*




The third pedal controls the noise produced by the key pad and is triggered when a key is pressed on the instrument. The level of the impulse noise is controlled by the closing speed of the key pad detected by the flute interface. This effect makes it possible to play the flute without blowing into it.

The fourth pedal allows the user to change the characteristics of the resonator of the instrument while the flute is being played. This means that we must design a filter allowing for large changes in the transfer function via few control parameters. We made a filter allowing continuous "morphing" between a flute-like and a string-like sound. By adjusting the filter's transfer function with a control pedal, interesting effects can be obtained. To obtain the filter, we first had to estimate the two transfer functions. These transfer functions, as shown in Figure 11, were obtained from measurements on a flute and a guitar string. Roughly speaking, they can be said to be low-pass filters with different cut-off frequencies and slopes. This suggests the use of a classical one pole filter with a z-transfer function as follows:

$$H(z) = G\frac{1-a}{1-az^{-1}}\ . \qquad (4)$$

A filter of this kind yields reasonable approximations of the two above mentioned transfer functions. In this way we can obtain interpolations between these transfer functions by jointly adjusting parameters $G$ and $a$, as shown in Figure 11.

In this application, the flute and Disklavier can interact in several ways. By adding a software layer to the MIDI interface, one can for instance use pressure variations from the flautist to control the speed at which the piano strings are struck, or control the tempo of a prerecorded sequence played by the piano.

## Conclusion

In this article, we have described a new interface that was adapted to a traditional flute. The goal in designing this interface was to give flautists access to the world of digital sounds without obliging them to change their traditional playing techniques. A synthesis model adapted to this interface was designed which makes it possible to resynthesize and transform the original flute sound. Because this synthesis model is a mixture between a signal model and a physical model, we have called it a hybrid model. Thanks to the physical model, we are now able to give the instrument a new set of physical characteristics, and thus give a physical interpretation of the sound transformation. This means that we can, for example, simulate the sound of a gigantic flute, or replace the resonator of the flute by the resonator of another instrument. As far as the signal model is concerned, its advantage is that it can simulate a sound no matter how complicated the physical processes underlying the source are. This means that when the physical behavior of a sound source is not fully understood, or when the physical equations which describe the source are too complicated to be implemented in real time, signal models can be used. These models generally yield a satisfactory sound resynthesis, and they enable one to perform sound transformations. In addition, this flute is MIDI-compatible, which means that it can be used to control other MIDI instruments, and can be played in any tuning system by assigning it arbitrary frequency values for a given key state.

This instrument has been presented twice to composers and musicians, and we are currently working with a composer to make a piece where the flute controls a MIDI piano. Traditional flute players who are interested in contemporary music have also given very positive feedback to this instrument, especially because they can use traditional playing techniques.

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