

Designing and Synthesizing Delay-Based Digital Audio Effects using the CORDIS ANIMA Physical Modeling Formalism

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Abstract — Throughout this paper, several CORDIS-ANIMA physical models will be presented to offer an alternative synthesis of some classical delay-based digital audio effects: a delay model, two comb filter models, three flanger models and a sound spatialization model. Several of these realizations support a control scheme based on the "Physical Instrumental Interaction". Additionally they provide several sonic characteristics which do not appear in the original ones. Especially the flanger model, for certain parameter values may give a new digital audio effect between flanging and filtering.

exceptionally simple digital signal processing building blocks such as adders and multipliers.

I. INTRODUCTION

A variety of digital audio effects (DAFx) make use of time delays. The echo, the comb filter, the flanger, the chorus and the reverb for example use as building block the time delay [1][2]. An evident digital synthesis-realization of the time delay is the digital delay line. Generally, the digital delay line and the unit delay is one of the few basic building blocks used in almost every audio digital signal processing algorithm.

However, when designing musical sound modification and sound synthesis units, a very important criterion, apart from the algorithm itself, is their control. In our digital audio effect designs, the "Physical Instrumental Interaction" - the type of physical interaction which a musician establishes with a real musical instrument- is fundamental [3][4][5]. Hence the synthesis of sound processing algorithms using structures that offer this type of interaction is the base of this research.

In the present article, after taking a brief look in the CA physical modelling and simulation system and its electrical analog, a number of digital audio effects models will be presented. A schematic block diagram will follow each model. These diagrams are combinations of CA networks with classical digital signal processing block diagrams; they describe in detail the sound modification algorithms.

II. CORDIS-ANIMA SYSTEM

A. CA network

In this essay, the proposed audio effects are actually computer models of physical objects. Moreover, several gesture models complete the "alphabet" from which the modification algorithms are designed. Generally, our toolbox contains a set of elementary virtual mechanical components -the CA modules (figure 1) - and

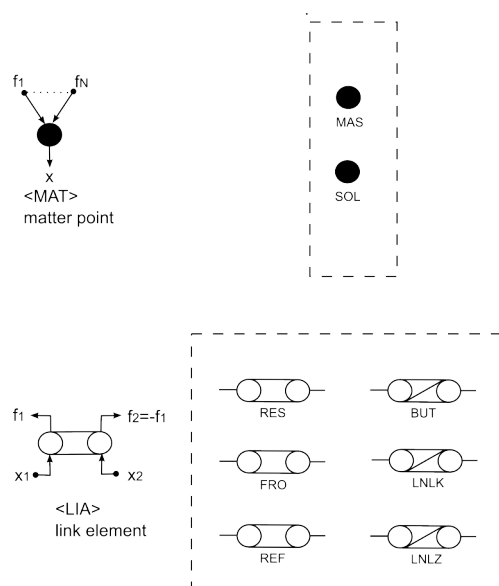


Fig. 1. CA modules

Each DAFx model is represented by a plane topological network whose nodes are the punctual matter elements <MAT> and links are the physical interaction elements <LIA> according to the CA formalism [6]. The simulation space used is limited to one dimension. Forces and displacements are projected on a single axis, perpendicular to the network plane. In tables I and II we depict the algorithms of CA modules. Table I provides the linear CA modules and table I the nonlinear CA modules.

These algorithms can easily take the form of ordinary signal processing block diagrams. However we prefer the CA network representation since its use makes the physical constitution of models easily perceptible and detectable.

In some models it was inevitable to enrich the CA networks with simple digital signal processing modules as adders, multipliers, switchers and filters. However the use of "externals" has been carried out with care in order to preserve the energetic passivity on certain parts of the models that was necessary.

TABLE I.
CA LINEAR ALGORITHMS

● MAS	$x(n) = 2x(n-1) - x(n-2) + \frac{1}{M} \sum_i^N f_i(n-1)$
● SOL	$x(n) = c$
—○—○— RES	$f_{RES}^{ij}(n) = K_{ij} \Delta x$ $\Delta x = [x_i(n) - x_j(n)]$
—○—○— FRO	$f_{FRO}^{ij}(n) = Z_{ij} \Delta v$ $\Delta v = [x_i(n) - x_i(n-1) - x_j(n) + x_j(n-1)]$
—○—○— REF	$f_{REF}^{ij}(n) = f_{RES}^{ij}(n) + f_{FRO}^{ij}(n)$

TABLE II.
CA NONLINEAR ALGORITHMS

—○—○— BUT	$f_{BUT}^{ij}(n) = \begin{cases} f_{REF}^{ij}(n) & x_i(n) - x_j(n) \leq S \\ 0 & x_i(n) - x_j(n) > S \end{cases}$
—○—○— LNLK	$f_{LNLK}^{ij}(n) = \text{interpolate}(\Delta x, T)$ lookup table $T : (f_r, \Delta x_r) \quad 2 \leq r \leq 20$ interpolation types : linear, cubic, splines
—○—○— LNLZ	$f_{LNLZ}^{ij}(n) = \text{interpolate}(\Delta v, T)$ lookup table $T : (f_r, \Delta v_r) \quad 2 \leq r \leq 20$ interpolation types : linear, cubic, splines

B. CA Electrical Analogs (Kirchhoff) : one-ports and 2-ports

An electrical analogous circuit of a mechanical system is an electrical circuit in which currents/voltages are analogous to velocities/forces in the mechanical system. If voltage is the analog of force and current is analog of velocity the circuit is called impedance analogous. In a similar way if voltage is the analog of velocity and current is the analog of force the circuit is called mobility analogous.

CA networks can be seen as a discrete time approximation of a subclass of mass-spring systems. Therefore electrical circuits may represent them easily. In practice to pass from a CA network to Kirchhoff network we transform in a first step the CA network into a mass-spring network. Then using the classical electro-mechanic analogies we obtain the Kirchhoff circuit [7].

The arrows represent the transition from one system formalism to another. We use a simple arrow symbol for the certain transition, a dotted arrow for the unsure transition and a marked arrow for the approximate transition. Table III gives the analogies between CA and electrical circuits variables and parameters.

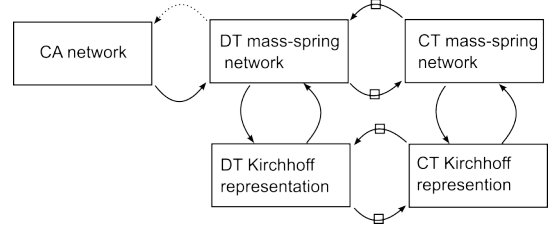


Fig. 2. From CA networks to discrete-time (DT) and to continuous-time (CT) Kirchhoff representation

TABLE III.
CA ELECTRICAL ANALOGS

Impedance Analog	Mobility Analog
$f \rightarrow v$	$f \rightarrow i$
$x \rightarrow q$	$x \rightarrow \varphi$
$M \rightarrow L$	$M \rightarrow C$
$K \rightarrow 1/C$	$K \rightarrow 1/L$
$Z \rightarrow R$	$Z \rightarrow 1/R$

III. DIGITAL AUDIO EFFECTS MODELS

A. CA Delay Model

A delay simply delays the input audio signal by an amount of time. For fractional sample delay lengths, interpolation techniques are used such as linear and allpass interpolation algorithms [1][2].

Synthesizing a delay line with the mass-interaction physical modeling scheme of CA is neither straightforward nor computationally effective. On the other hand, its algorithmic structure can be interesting as it offers a mentally strong physical metaphor and permits directly a control based on the "Physical Instrumental Interaction".

From figure 2 we observe that often we can pass from a continuous-time Kirchhoff network to a CA one. Luckily an electrical transmission line, which may be considered as an analog delay line, can be transformed to CA structure. Hence, in CA system, a digital delay line takes the form of virtual string terminated by its characteristic impedance.

In figure 3 we depict the network of an electrical delay line and its CA realization. The impedance analog network has been used. The stiffness parameter K of the model controls the time delay. Analytic expressions of the time delay as a function of the model will be presented later on.

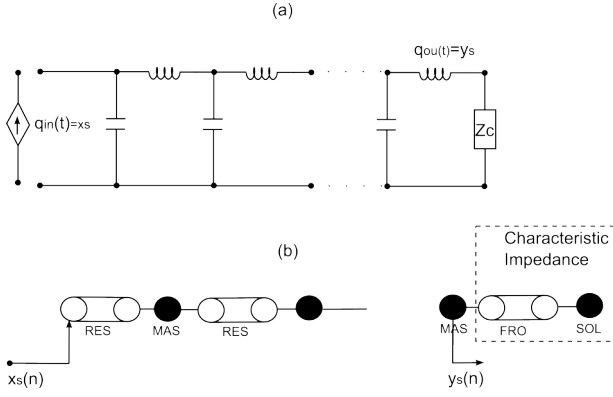


Fig. 3. (a) electric transmission line (b) CA delay model (x_s : input sound, y_s : output sound)

We can compute the characteristic impedance by expressing and decomposing the CA model into two-ports. An approximate analytic expression of the time delay as a function of the model parameters can be computed as well by the same decomposition. Figure 4 illustrates an elementary CA two-port used. Equation (1) expresses mathematically its input/output terminal relations.

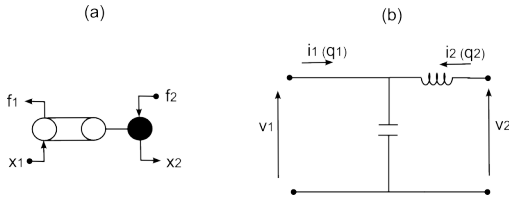


Fig. 4. From CA networks to discrete-time (DT) and to continuous-time (CT) Kirchhoff representation

$$\begin{pmatrix} f_1 \\ f_2 \end{pmatrix} = \begin{pmatrix} K & -K \\ K & -K - M \frac{1 - 2z^{-1} + z^{-2}}{z^{-1}} \end{pmatrix} \begin{pmatrix} x_1 \\ x_2 \end{pmatrix} \quad (1)$$

The concept of applying the of two-ports representation to the digital signal processing domain is not new as it has been introduced earlier in a similar way by Mitra [8]. However in the CA formalism it has never been used before.

The terms of the matrix that appears in the equation (1) are called impedance parameters z_{ij} [9]. The characteristic impedance Z_c and the delay time D are functions of those parameters [10]. Equations (2) and (3) give their analytic mathematical expressions (we express the impedance parameters by their Fourier-transform). The total time delay for a N -MAS CA delay line is given by the equation (4).

$$Z_c = K - K \frac{2Ke^{-j\omega} + M(1 - 2e^{-j\omega} + e^{-2j\omega})}{2Ke^{-j\omega}} \quad (2)$$

$$\pm K \sqrt{\left(\frac{2Ke^{-j\omega} + M(1 - 2e^{-j\omega} + e^{-2j\omega})}{2Ke^{-j\omega}} \right)^2 - 1}$$

$$D = \frac{d \cos^{-1}(\lambda)}{d \omega} \quad \text{with } \lambda = \frac{Z_{22} - Z_{11}}{2Z_{12}} \Rightarrow \quad (1)$$

$$D = \frac{d \cos^{-1} \left(\frac{2Ke^{-j\omega} + M(1 - 2e^{-j\omega} + e^{-2j\omega})}{2Ke^{-j\omega}} \right)}{d \omega} \quad (3)$$

$$D_{\text{total}} = ND \Rightarrow$$

$$D_{\text{total}} = N \frac{d \cos^{-1} \left(\frac{2Ke^{-j\omega} + M(1 - 2e^{-j\omega} + e^{-2j\omega})}{2Ke^{-j\omega}} \right)}{d \omega} \quad (4)$$

We observe from the last expression (equation (4)) and we also perceive from the audio outputs that the model suffers from dispersion. It is remarkable that for $M=K=1$ certain values the CA delay line synthesizes precisely the time delay without phase distortion or undesired filtering. The Z_c and D in this case are:

$$Z_c = 1 - z^{-1} \quad (5)$$

$$D = 1 \text{ and } D_{\text{total}} = N \quad (6)$$

The characteristic impedance expressed by the equation (5) can be synthesized in CA by a FRO module with $Z=1$ attached to a SOL module.

Instead of the precise derived equations (2)-(4) we may use their approximation in the continuous-time domain as given by the continuous-time electrical network. Using the results from the electrical transmission lines [10] and the CA-electrical analogs we get the following helpful approximations:

$$Z_c = \sqrt{MK} \quad \text{for } \omega^2 \frac{M}{K} \ll 4 \quad (7)$$

$$D = \sqrt{\frac{M}{K}} \quad \text{and } D_{\text{total}} = N \sqrt{\frac{M}{K}} \quad (8)$$

B. CA Comb Filter Models

A comb filter is produced when a slightly delayed audio signal is mixed with the original one [1][2][11]. When the delayed version is fed back to the delay line input we have an IIR comb filter. Otherwise we get an FIR comb filter. Both topologies give a large number of notches in the spectrum of the input signal.

Two models that synthesize this classical digital audio effect are depicted in figure 2. The first one uses the = delay model. A similar effect to the IIR one is experienced naturally inside an acoustical cylinder when a sound circulates inside it: the successive reflections at both ends of a cylinder modify the signal approximately as a IIR comb filter. It is not difficult to simulate this phenomenon with a CA string model (second model of figure 2). The resulted effect is perceived as a natural resonator. Two important differences from the signal processing model are 1. the notches do not cover the whole spectrum (the

number of notches are defined by the number of masses) and 2. the dispersion is inevitable in contrast to the first one where it is inexistent.

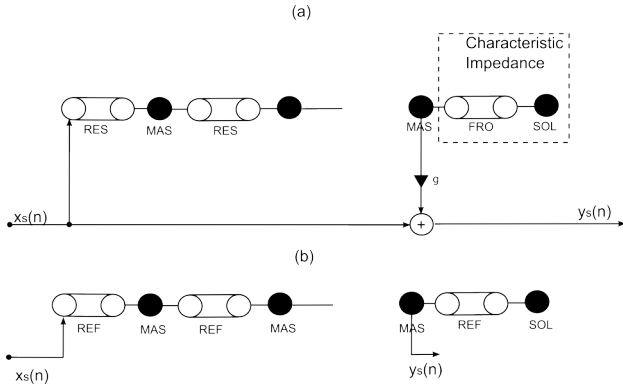


Fig. 5 (a) CA comb filter model using the delay model (b) CA comb filter model using a CA string (x_s : input sound, y_s : output sound)

C. CA Flanger Models

A flanger can be easily implemented using variable length delay lines [1][2][12][13][14][15]. Basically it is a comb filter where the delay line length is slightly changed. Hence we may use the previous CA comb filter models where the parameter K that defines the time delay approximately according to equation (8) is altered periodically (figure 6a). For wide amplitude modulation values of the K parameter, a new effect is obtained between dynamic filtering and flanging.

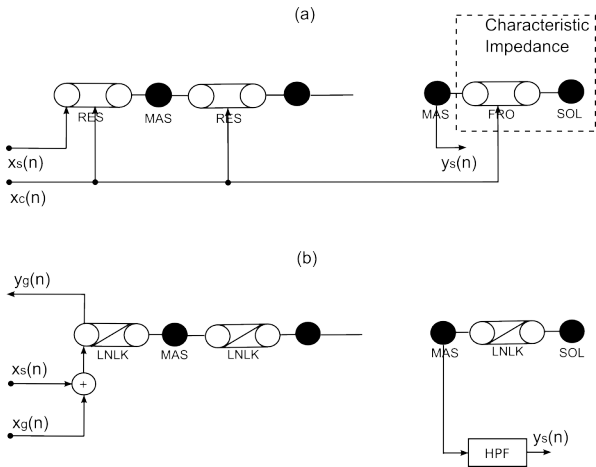


Fig. 6 (a) CA flanger model using a comb filter model (b) CA flanger model using a nonlinear CA string (x_s : input sound, y_s : output sound, x_c : control input, x_g : gesture input, y_g : gesture output)

A more physical approach in the context of “Physical Instrumental Interaction” is to use a variation of the second CA comb filter model where the linear <RES> modules are exchanged by non-linear <LNLK> modules. The designed non-linearity characterizes the time delay. A gesture stressing the physical model biases it and consequently determines the linear regions of the system. Therefore this gesture affects the time delay of the comb filter structure.

Another flanger model will be presented in the next section. It has been preferred to be described in that part of the paper because of the adopted design approach.

D. CA Spatialization Model + Flanger Model

The CA networks have an inherent spatiality due to their topology. The sound can be picked-up from every elementary CA basic module output. Figure 7 represents a simple CA flanger model with two outputs. Due to the relative time delay between the two output nodes in the network, we obtain a spatial image of the sound source (the interaural time differences (ITD) are a strong cue that the human auditory system uses to estimate the apparent direction of a sound source [1]). Hence the geometrical spatial characteristics (a more accurate term would be topological) of CA models are quite related to the spatial sound characteristics of the outputs.

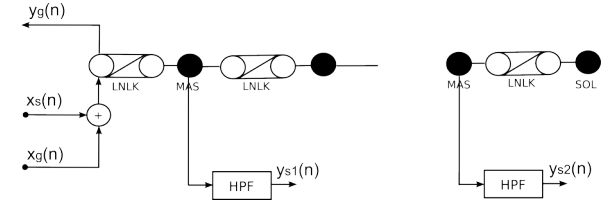


Fig. 7 (a) CA spatilization model (x_s : input sound, y_s : output sound, x_g : gesture input, y_g : gesture output)

The analysis of the chapter III.A can be applied in order to choose the proper CA nodes to obtain a desired spatial image of the sound source. It is clear that the spatial discretization quantizes the spatial trajectories. The interaural intensity differences (IID) can be used as well to improve the spatial image.

The use of movable pick-up points gives the opportunity to attain dynamical effects. A similar idea has been applied earlier to digital waveguides [16]. The CA model of figure 8 is a type of flanger. Each pick-up point determines the partials reinforced by the string topology. If we place the pick-up point at a position $1/m$ across the sting length, the partials whose number is m will be canceled.

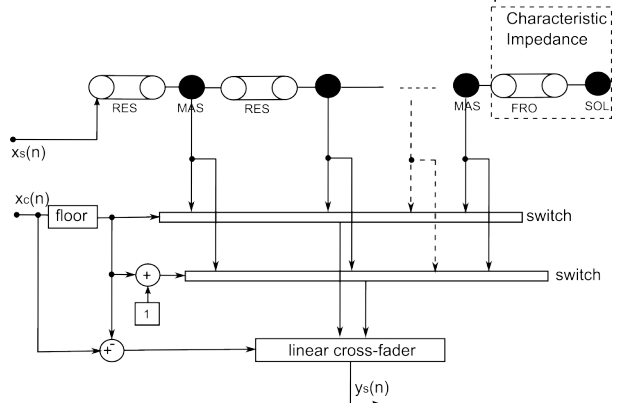


Fig. 8 CA flanger model using a pick-up point modulation (x_s : input sound, y_s : output sound, x_c : control input)

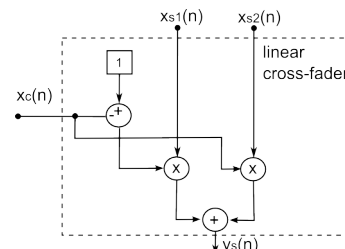


Fig. 9 Linear cross-fader block diagram

IV. FUTURE WORKS AND CONCLUSIONS

All of the above models were simulated and run in different times. Several other simple models were designed to simulate the physical mechanical gestures used to control and interact with them. In the near future all of them and several others used as well for sound processing will be implemented in real time and controlled by force feedback haptic interfaces as the TGR developed in ACROE laboratory [17].

The aim of this general research is to develop new tools for sound transformation mainly used for musical purposes that preserve the important natural instrumental relation found in acoustical musical instruments. We believe that through this physical dynamic control of the audio effect process, a virtuosity will emerge that will contribute to the quality and the finesse of sound transformation.

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