Digital music synthesis: A teaching tool for Signal Processing Courses

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Abstract: - Digital Signal Processing has many applications that are very interesting to undergraduate students. One way to put emphasis in important concepts is through Digital Music Synthesis. The paper will show how to exercise and master some aspects of signal processing using virtual music instruments. Our results are very encouraging, awakening student interest in DSP.

Key-Words: Digital Music Synthesis, Signal Processing Education, Physical modeling

1 Introduction

In our country, engineering is a five-year career. The general idea is to provide the future engineer a wide background, in order to increase his chances in the job market.

In this context, we work as full-time professors who teach Signal Processing for senior undergraduate students. We are the heads of the DSP Laboratory, which, although quite small, is always active [1].

We think that laboratory hands-on experiments are the best way to learn, and certainly more fun. Among several subjects we get involved, computer music has a special effect. Most students go far beyond what they are asked because of their high motivation.

To introduce such issues in our courses, we acquired proper background in the subject after some reading and several years of practice. MATLAB [2] is our companion tool and recommended references for beginners are [3], [4] and [5].

Students are prepared for the task given that they had a course on Signal and Systems, and they are very familiar with MATLAB programming.

2 Concepts involved

2.1 Time frequency relationships

Study of discrete time signals includes DTFT (Discrete-Time Fourier Transform) and DFT (Discrete Fourier Transform). Traditional SP courses go deep into such topics. But full understanding of processed music is not complete without the study of spectrograms. We also think that spectrograms are a good tool to practice DFT concepts and to introduce filter banks. We profit from the MATLAB function `specgram`, which the students must master. To reach to DFT and time limitations, they play the overlapping parameter and different windows.

A signal displayed as a spectrogram is a way to understand time-frequency relationships and displaying limitations. Figure 1 shows these results using a Hamming window and different overlapping values. A trade-off between frequency precision and tine definition is quite clear.

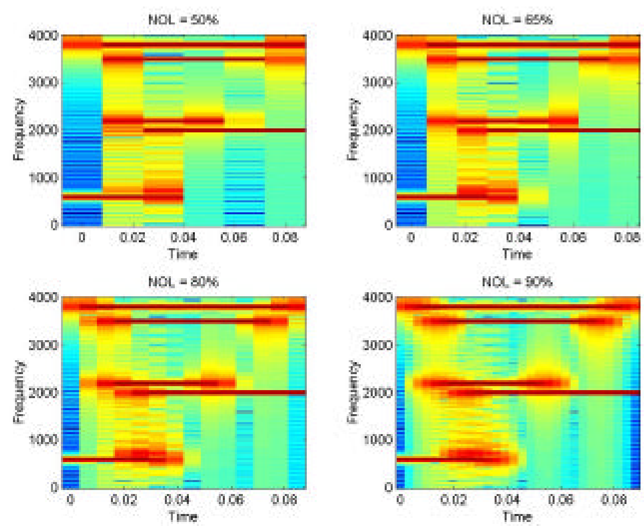

Fig. 1

A second step to build the knowledge base is the envelope of a single tone audio sequence. In order to represent a musical note played by a certain instrument, its shape should be similar to one of the signals showed in Fig. 2.

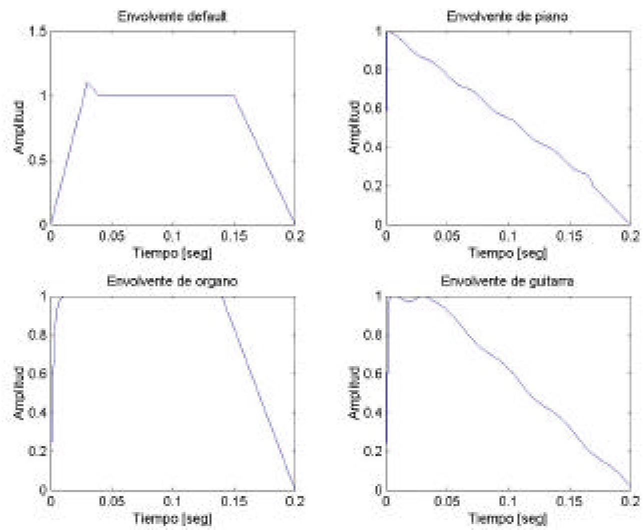

Fig. 2

The effects of the first signal of Fig. 2 modulating a pure tone can be seen in the spectrogram of Fig. 3.

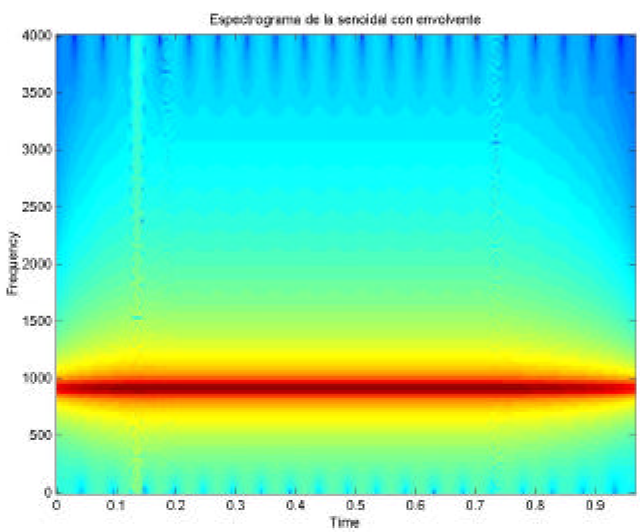

Fig. 3

Using MATLAB function sound, this effect can be heard and appreciated.

To grasp the idea of digital music, we begin with additive synthesis.

We sum a harmonic set of frequencies which have different amplitudes and similar, but not equal, envelopes. Several parameters allow us to trim the sound in order to have a virtual instrument. These items are: frequency, number of harmonics, attenuation law of the harmonics, time frame of each harmonic and envelope shape. It can be seen that the amount of variables is very high, allowing results that vary from fairly good sounds to fearful shrieks.

Once we get an acceptable sound, the creation of a melody is quite simple. We add signals altogether when we want accords and we superimpose them in time, the way musicians play a piano. Fig 4. displays the spectrogram of the melody "Ode to Joy" with the considerations described above.

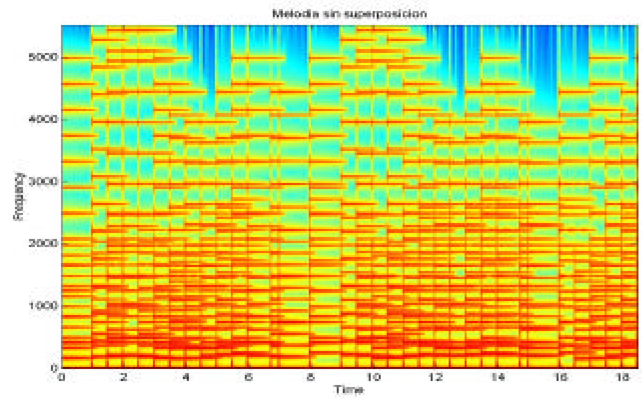

Fig. 4

2.2 Transfer Functions and Frequency Response

When we are concerned about showing students applications where H(z), z-plane representations and stability, we may introduce physical instrument synthesis.

The Karplus and Strong model of a plucked string and further studies on the subject [6] [7] [8], let us work with all the concepts at the same time. But what is better, students can "hear" the effects of any change. It is not the purpose of this paper to explain how the system works, but to point out relevant topics that are involved in physical modeling. The interested reader can go deep in the subject following the references.

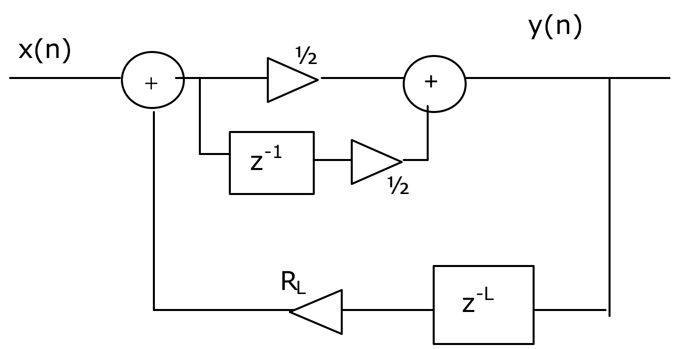

Fig. 5

The KS model of Fig. 5 responds to the following transfer function:

$$H(z) = \frac{z^L(z+1)}{2.z^{L+1} - R_L.z - R_L} \quad (1)$$

Where L is a parameter related to the note frequency given by (2) and R_L gives the persistence of the sound.

$$\boxed{f_o = \frac{f_s}{L + 1/2}} \qquad (2)$$

Using the simple approach of [6] the infinite impulse response will decrease to the 10% of the initial value after:

$$t_D = \frac{1}{f_o}\frac{\ln(0.1)}{\ln(R_L)} \qquad (3)$$

It can be shown that a better control of this time can be achieved using a FIR filter in the feedback loop.

$$t_D = \frac{1}{f_o}\frac{\ln(0.1)}{\ln\left[R_L . \cos\left(p\,\frac{f_o}{f_s}\right)\right]} \qquad (4)$$

In this case the frequency of the note will also be modified. A plot of a typical frequency response is shown in Fig. 6 and its corresponding z-plane pole-zero plot in Fig. 7.

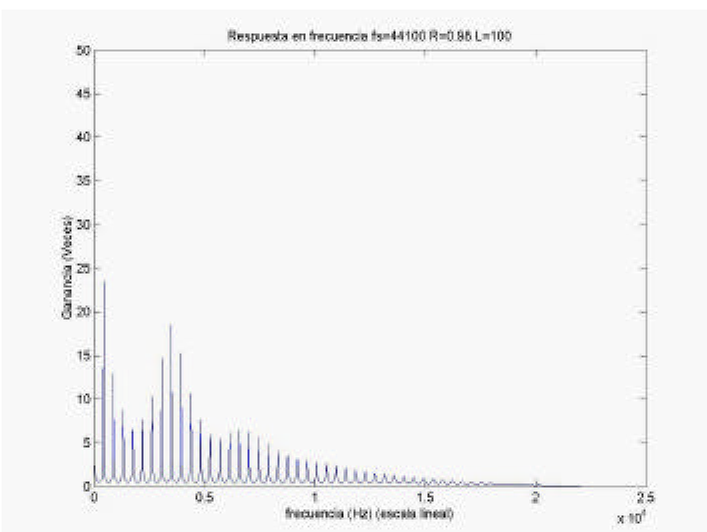

Fig. 6

The sound can be further improved in many ways. A high pass filter in the loop to take care of the inconvenient DC effects can be added [8]. We also studied the inclusion of an all-pass filter [7] to achieve an exact match in frequency without annoying phase errors. There is a way to smooth the sound with a moving average pre filter that can make a difference in some frequency range. Fig. 8 shows a simplified block diagram.

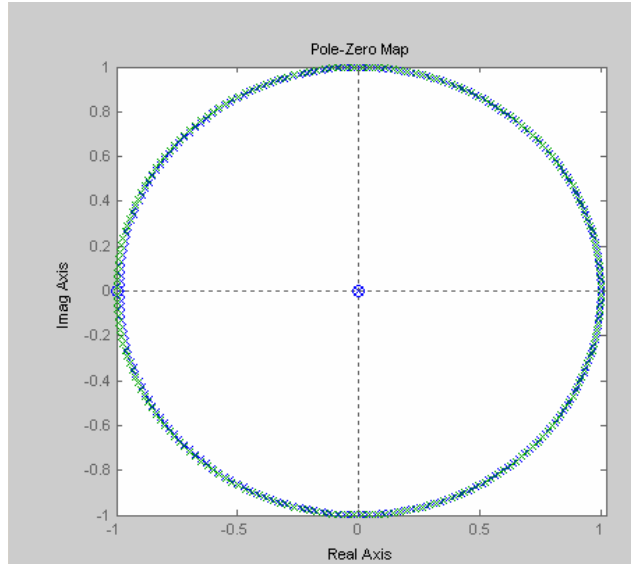

Fig. 7

But the election of order and proper coefficients, or even the inclusion of each block, has been the result of many experiments and deeper research [9][10].

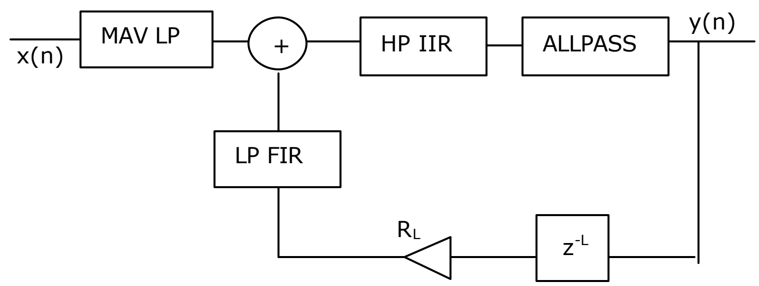

Fig. 8

Stability is always a matter of concern in a feedback system like this one. Students must acknowledge the problem each time they introduce a variable in the loop. It really helps understanding the concept, because an unstable system implies a "noisy" output.

3 Results

One of the most rewarding experiences regarding this subject is student assessment.

High motivation and easily comprehensive outcomes help us reach very good programs.

They usually have a MATLAB GUI interface where the user can choose among many virtual instruments, generated from different synthesis approaches. They also include melodies they get from midi files, but that are played by a self-tuned guitar or a "weird" organ. Fig. 9 is an example of a front end.

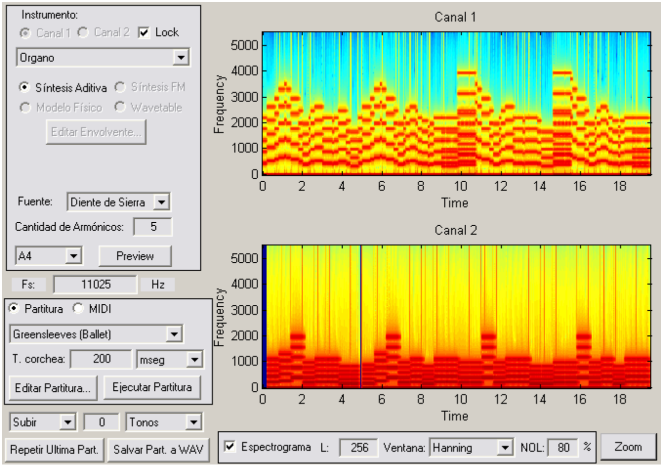

Fig. 9

Although it is not mandatory, they build up a Simulink model of a plucked string, like the example of Fig. 10, where the mask of the modified algorithm of [7] may be tested.

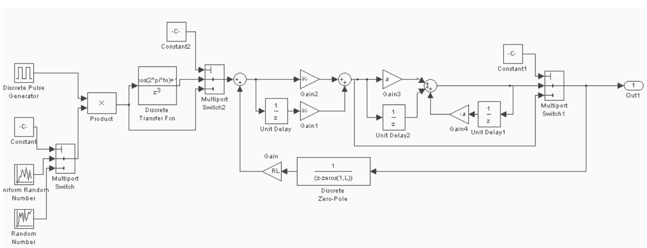
Fig. 10

In order to improve the quality of sounds, they also implement wavetables. They download recorded sequences of real instruments and they perform the rest of the processing (envelope and resampling).

We introduced synthesis in our curricula since 1998. From then on, we have collected percentage of acceptance based on assignment response (see Fig. 11). Although there could be other factors that may be correlated, we think that results worth the efforts involved.

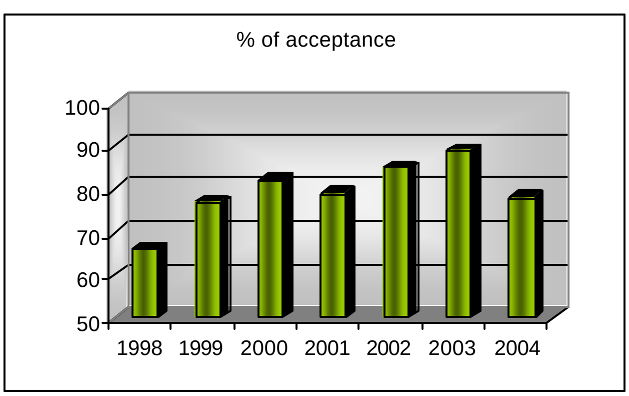

Fig. 11

4 Conclusions

Our proposed goal was to motivate students to learn signal processing. The fact that they ended up producing digital music was a plus. In between we could reach to them covering many aspects of the curricula, such as:

- DFT Processing and Spectrogram understanding
- Comb filtering
- White noise processing
- Loop stability
- Phase equalization
- Low-pass filtering, to emulate in-instrument absorption of sound waves
- MATLAB and Simulink's capabilities
- Research skills

Theory and applications work perfectly fine together to develop SP student knowledge. We believe that digital music is a very good example.

Acknowledgements

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