Six Fantasies Machine (SFM) Version 0.50 – Manual

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1.0 General description and overview

The SFM instrument is a tool developed in *csound* for producing a range of sounds similar to those found in Paul Lansky's classic computer music composition *Six Fantasies on a Poem by Thomas Campion*. This piece was made in 1979 using dominantly LPC, linear predicative coding, and comb filters. The piece was made, with some exceptions, from a sound recording of Lansky's wife, Hannah MacKey reading Campion's poem. The five first movements in the piece each explore different aspects of the voice and different techniques. LPC is an analysis/resynthesis technology, promising at the time of composition, that calculates the filter coefficients from a speech as well as the fundamental frequency and the amplitude of the speech signal. It can be used in combination with a buzz and a noise source controlled by the parameters for analysis to produce a speech signal. Through this technique the fundamental frequency, the spectral envelope, the pitch values and the duration can be controlled independently from each other. Lansky also used double comb filters extensively in the third and fifth movements, where he put together banks of comb filters that could produce complex spectra.

The analysis files for SFM was made by recording the voice of actress Nancy Helms imitating Hannah MacKey's reading that is presented in the last of the fantasies. Five recordings were made and the best individual phrases were selected. Due to recording levels set to low the resulting sound files had to be increased in gain by 17.3dB's. Then, the sound files were resampled and analyzed using lpanal, the LPC analysis utility in *csound*. These files are provided in the instrument.

SFM has eight voices that can be manipulated individually. The resulting sound can be written to sound files if desired. Thus, the instrument can be used as an exploration of sounds in the footsteps of Lansky, and as a tool for composition.

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2.0 Installing and running

2.1 Install csound

Install csound on your machine if you haven't already installed it. The program can be downloaded from http://csound.sourceforge.net/#Downloads . You should install the float instead of the double version (The current version of SFM is made with Csound5.10.1-gnu-win32-f.exe).

You might be told by the installer to install Python2.5. This should not be necessary: but if you want to ensure that csound runs smoothly and steadily you might consider doing this.

1.2 *Unzip*

Unzip the Six_Fantasies_Machine.zip to the directory where you would like to locate the program.

1.3 *Create a shortcut and run the program*

If you want: create a shortcut to the program that you can place on your desktop by right clicking the icon and choosing **Create shortcut**.

Open the folder which the unzip program created. Run the program by double clicking the icon marked **Six_Fantasies_Machine-Win**.

3.0 Playing and writing sound files

3.1 Playing sound

If your soundcard is properly connected and you have sound on your system you should hear a sound when the program opens.

Pressing the **"PLAY"**-button would thereafter produce sound according to the settings applied with the interface.

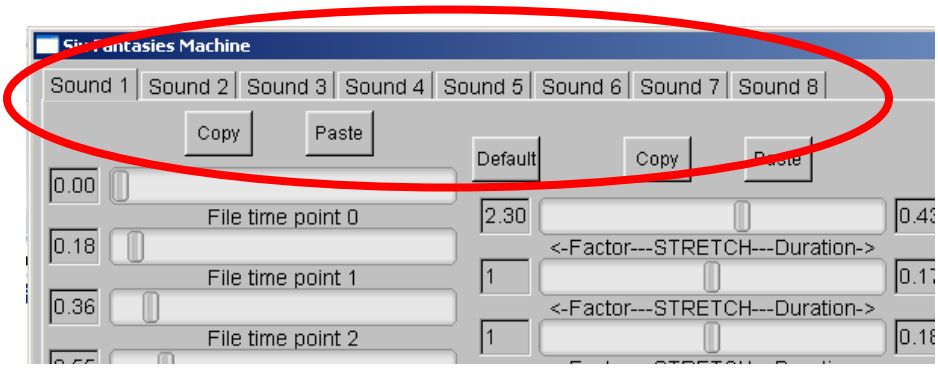
3.2 Writing soundfiles

You can write sound files to disk by pressing the **"Write file"**-button. This will produce a file with the name SixFantasiesSound1.wav to the **/snd** folder. Pressing the button again will subsequently produce a file with the same name but with increasing numbers: SixFantasiesSound2.wav: etc. If you start a new session: the numbers will start at 1 again: so if you want to keep sounds from an earlier session: you should rename them or move them to another folder.

NB! As it was for Paul Lansky, the sound files written by this instrument has a sampling rate of 14kHz.

4.0 Settings for the individual voices

There are eight tabs on the interface corresponding to the eight voices in the instruments. By clicking on a tab one will switch to the sliders, buttons and counters corresponding to the corresponding voice.



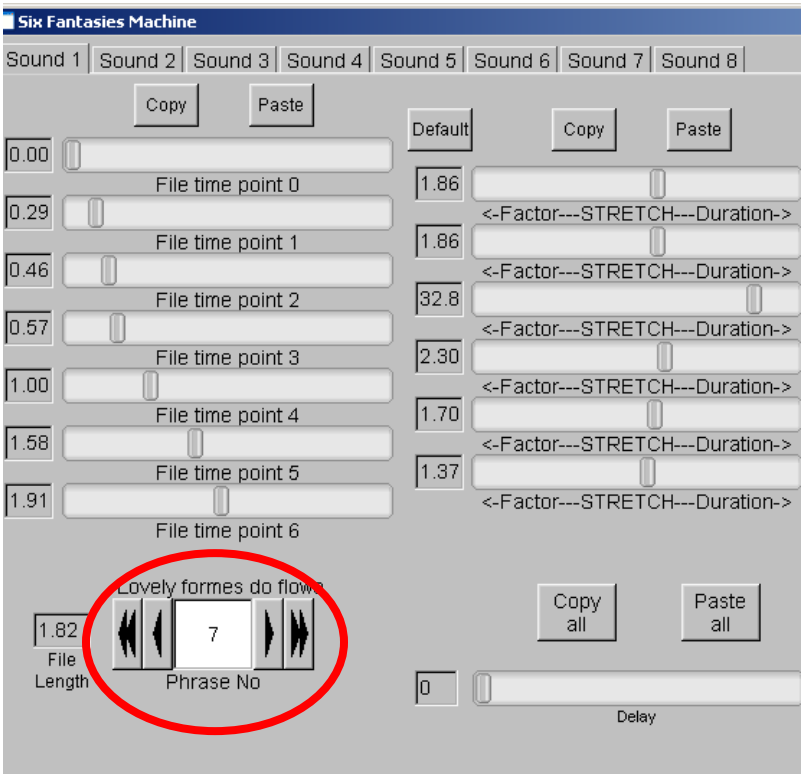
4.1 Phrases

Each of the phrases in the poem has been numbered thusly:

1: "Rose cheeked Lawra"
2: "Come"
3: "Sing thou smoothly with thy bewties"
4: "Silent Musick"
5: "Either Other"
6: "Sweetly Gracing"
7: "Lovely formes do flowe"
8: "From Consent Divenely Framed"
9: "Heaven is Musick"
10: "And Thy Beawties Birth"
11: "Is heavenly"
12: "These dull notes we sing"
13: "Discords Needs for Helps to Grace Them"
14: "Only Beawty Purely Loving"
15: "Knows no discord"
16: "But still"
17: "Moves Delight"
18: "Like Clear Springs Renued by Flowing"
19: "Ever perfect"

20: "Ever in Themselves"
21: "Eternall"

By using the counter at the lower left part of the tabs one can set which phrase one would like to play:



By clicking on the right arrow one will choose the phrase with a higher number, and by choosing the left arrow one will choose the phrase with the lower number. The double arrows will increase or decrease the phrase number by 4.

By choosing one particular phrase one will at the same time automatically show the phrase duration and set the File Time Point sliders at points evenly distributed in the phrase.

4.2 File time points

There are seven file time points that can be set with sliders for each phrase. Each file time point constitutes together with another point with a higher or lower number segment of the phrase which can be stretched, compressed or given a particular frequency or pitch value. E.g. if File Time Point slider 1 is set to 0.00 and File Time Point slider is set to 0.15, this will constitute a segment from the start of the file and 0.15 s. into the file. This means that this particular segment can be stretched with the STRETCH slider or given another pitch with the pitch slider.

Together the File Time Points constitute six segments that correspond to the six STRETCH and Pitch sliders.

The STRETCH and pitch sliders that corresponds to each segment is placed on a imaginary horizontal line running from between the File Time Point slider and to the right on the interface.

One needs not set the file time point so that they comprise the whole phrase.

NB! The values for the seven file time points should be set in increasing order. Setting them otherwise might cause unexpected behavior!

4.3 Stretching and compressing segments

There are six sliders for setting the STRECTH factor for the corresponding segment. Moving the slider to the right increases the duration of the segment, while moving it to the left decreases it. Thus, one can both stretch and compress segments.

The numbers showing up in the small boxes to the left of the sliders shows the stretch factor for each segment, where a value of 1 gives the original duration, 2 gives the double duration, 0.5 gives the half duration, and so forth. The small boxes to the right of the slider shows the duration of the segment. Beneath all the duration boxes there is a box showing the total duration of the phrase in its stretched or compressed state.

By clicking on the “Default” button, all the sliders are set to 1.

4.4 Delay

The delay slider sets the delay in seconds for the onset of the voice in question. If on wants to organize a sequence of sounds played on different voices this is an essential feature.

4.5 Manipulating the pitch of the phrases

There are several different ways of setting the pitch value and behavior in SFM. The two main sources producing pitch values are

1. The original analysis file.
2. The pitch sliders to the right of the interface.

4.5.1 Choosing pitch source

By adjusting the “Pitch source” slider one chooses between either having pitch from the analysis file (value 0) or setting pitch from the six pitch sliders (value 1). Values between 0 and 1 give a weighted average of pitch values from the analysis file and the pitch sliders where 0.5 implies equal weighting of the two.

4.5.2 Range

The “Range” slider only has effect on the pitch values from the analysis file. The range value is calculated on the basis of the average pitch value of the whole phrase, where the range value multiplies/scales pitch values that deviate from the average with the following implications:

- 1 => no change
- 0 => flat pitch contour at average pitch value
- 2 => Deviations from the mean are multiplied by 2 and thereby exaggerated
- -1=> The pitch contour is inverted around the average value
- -2=> The pitch contour is inverted and the deviations are multiplied by 2.

4.5.3 Transposition

Transposition applies to changes in pitch using any of the methods described, that is, on pitch contours that are set from the analysis file or from sliders, or that may be manipulated in terms of range.

The slider sets transposition in number of semitones and 10*cents, so that a value of -6.2 will imply a transposition downwards of a tritone plus 20 cents.

4.5.4 Setting pitch from sliders

If the pitch source is set to 1 one can control the pitch of the segments defined by the File Time Point sliders. The pitch value that is set for one segment will apply for the whole segment. However, there will be a gliding transition (protamento) towards the subsequent pitch value if this is different.

The speed of the transition between subsequent pitches can be set with the "Portamento" slider. Small values indicate a short transition. The default value is 0.1.

The number boxes to the left of the sliders show the frequency value of each segment, whereas the right value shows that corresponding note name. + and – are indicate deviations from equal temperament in either direction. The different pitch names are all associated with a unique color.

By clicking the "Mean" button all the pitch sliders will automatically be set to the average value.

By clicking the "Tune" button, the frequencies of all the pitch slider will be adjusted to the nearest note in the tempered scale.

4.5.5 Pitch or noise as excitation

By changing the values of the "Noise/buzz" slider, one can choose between having

- Buzz only (a value of 0) except the parts that are noise excited in the analysis file, for instance all unvoiced consonants.
- Noise only (a value of 1). All sound will be noise excited.
- A mixture of buzz and noise, where smaller values increase the degree of buzz and increase the value of noise and larger values do the opposite.

4.6 Shift spectral envelope

The spectral envelope of the filter part (the hills and dale one can imagine creates the "outline" of all the frequencies in the spectrum) of the analysis file can be manipulated by using the "Spectral shift" slider. The values indicate a scaling value applied to the frequency envelope. Here are some examples:

- 1=> No change
- 2=> The spectral envelope is shifted upwards by a factor of 2. A spectral peak occurring at 1200Hz in the analysis file will lie at 2400Hz after being manipulated.
- 0.5=> The spectral envelope is shifted downwards by a factor of 0.5. A spectral peak occurring at 1200Hz in the analysis file will lie at 600Hz after being manipulated.

The spectral envelope of voices are an important cue of gender and age. Men usually have spectral envelopes about 20% lower than women (corresponding to a value of 0.8

in SFM). Children can have spectral envelopes of 20% above that of women (corresponding to a value of 1.2 in SFM).

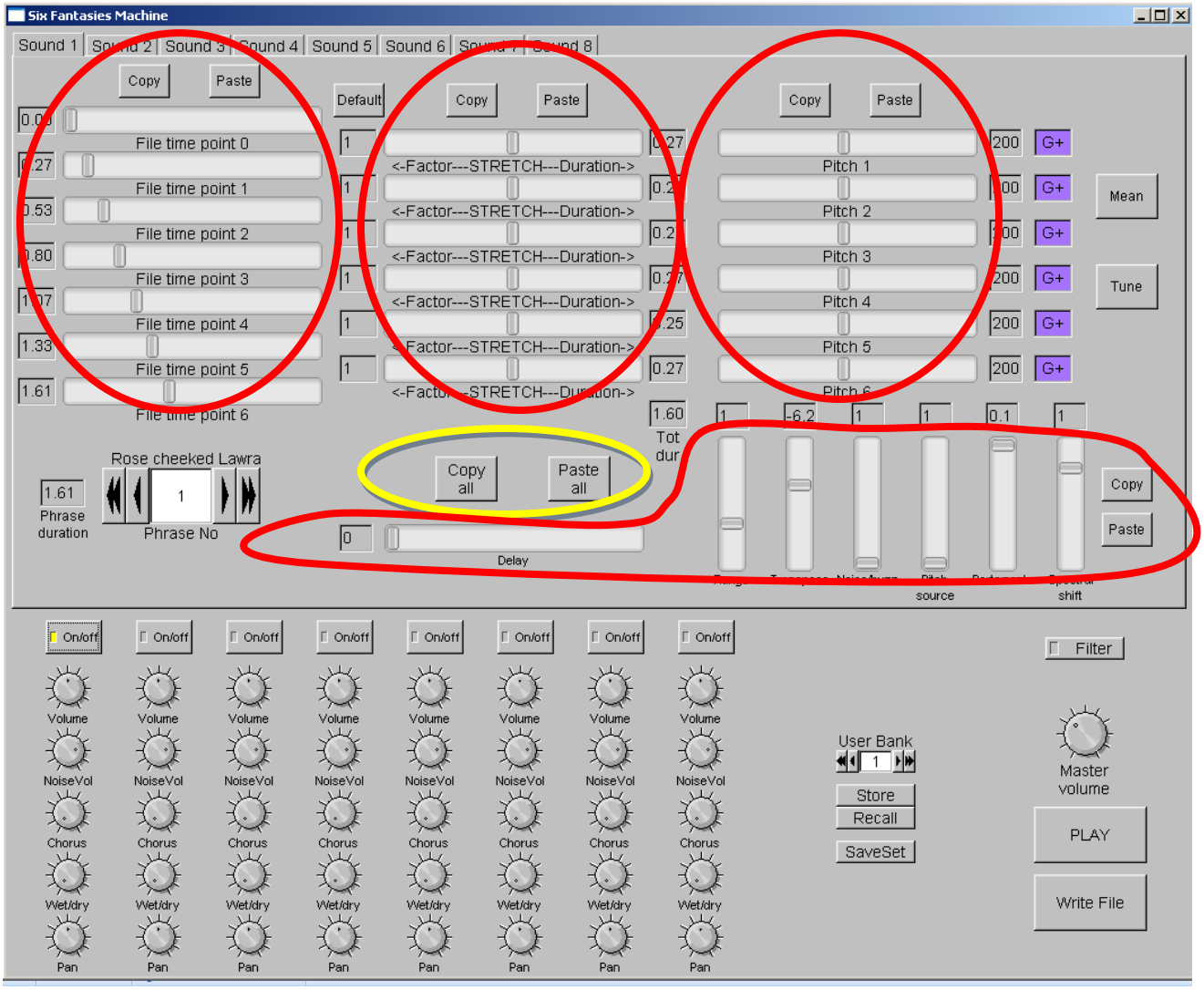
4.7 Copy and paste

Each of the tabs in SFM are equipped with five sets of “Copy” and “Paste” buttons. These are for copying values from one voice to another, a feature that can be convenient if one wishes to have several voices with related behavior.

Four of these pairs are associated with a particular set of sliders, marked on the illustration with red ellipses / areas.

One pair are global copy/paste buttons. This is marked with a yellow ellipsis in the figure.

The phrase counter remains unaffected by any of the copy/paste buttons.



5.0 Voice controls

In the lower area one can set a number of parameters for each of the eight voices in the instrument. The controls are from top to bottom:

1. **On/off:** Turning the sound in the voice on and off.
2. **Volume:** Volume of the individual voice.
3. **Noise Vol:** Volume of the unvoiced parts of the analysis files like unvoiced consonants ([h], [s], [f], etc.)
4. **Chorus:** “Chorus”, that is, really the amount and number of superimposed versions of the buzz source with random pitch modulation that is mixed before filtering. When the “Chorus” knob is set to maximal, four additional buzz generators, each with different

random pitch variation are mixed with the main buzz source before filtering. This causes a markedly richer timbre.

5. **Wet/dry:** Reverberation wet/dry mix. Low values means little reverberation, high values means a high portion of reverberated sound.
6. **Pan:** Panning between left and right channel.

6.0 Using filters

As in Lansky's *Six Fantasies* one can also work with filters in this instrument. The filters are activated by checking the "Filter" button at the right of the lower part of the panel. This automatically loads a set of parameters for controlling the filters specified in the file named "Filter.txt" placed in the /flt folder.

The parameters of the filters can be edited by importing the "Filter.txt" file into a spreadsheet or by using a simple text editor such as notepad. The parameters are organized so that one line of text just above the parameters indicate what they are. The eight columns (or values, in a text editor) contain parameters for each of the eight voices.

The filters can be edited while running the instrument. To load new filter settings, save the "Filter.txt" file, then uncheck and check the "Filter" button.

There are nine filters per voice, giving a total of 72 filters. The parameters that can be controlled are:

6.1 Mix:

The mix between dry and filtered sound is set in the first row of numbers. A value of 0 indicates a dry setting (=no filtering), and a setting of 1 will produced only filtered sound. This parameters thus controls the output of all the filters together.

6.2 Type:

Three values are allowed:

- 0 – double comb filtering
- 1 – band pass filtering
- 2 – a combination of band pass and double comb filtering. The order is band pass -> comb

6.3 Pitch

This parameter sets the pitch produced by the double comb filter (that is, if it is in the audible range) or the centre frequency of the band pass filter. One can choose between

setting the values as frequencies or as pch-values. Any values below 20 will be interpreted as values in the pch format.

The pch format makes it possible to use tempered pitches. The format is *octave.pitchclass*. *Pitchclass* give the pitches from C to H as decimal values between .00 and .11. Thus an A would be .09 and an E would be .04. The octave value is given as an integer so that an A of 440Hz is given as 8.09.

6.4 Decay Time

The decay time parameters only have an effect for comb filters (mode 0 and 2). They refer to the time it takes for the signal to decay to 1/1000 of its value. Actually, since the filtering is double the setting refers to *half* the time in seconds. Adjust according to preference.

6.5 Band Width

This parameter only has effect for the band pass filter modes (1 and 2). They refer to the width in Hz between the lower and higher cutoff (-3dB) points.

6.6 Gain

Filtering can radically affect the intensity of the signal. Therefore it is often necessary to adjust the gain of the filtered signal. To prevent overflow the gain values for the comb filter modes (0 and 2) are multiplied by 0.001 compared to that of the band pass mode (1).

Tip: If one needs more than nine filters to produce complex resonances, one can use several voices with the same settings and apply filters for the corresponding voices. Thereby, one can in principle make a sound with 72 different resonant components.

7.0 Using presets

A bank of 50 presets can be accessed through the “Presets” counter in the right lower section of the interface. By clicking the “Recall” button one can set the instrument according to the presets.