Jean-François Charles
1 Route de Plampéry
74470 Vailly, France
jfc@jeanfrancoischarles.com

A Tutorial on Spectral Sound Processing Using Max/MSP and Jitter

For computer musicians, sound processing in the frequency domain is an important and widely used technique. Two particular frequency-domain tools of great importance for the composer are the phase vocoder and the sonogram. The phase vocoder, an analysis-resynthesis tool based on a sequence of overlapping short-time Fourier transforms, helps perform a variety of sound modifications, from time stretching to arbitrary control of energy distribution through frequency space. The sonogram, a graphical representation of a sound's spectrum, offers composers more readable frequency information than a time-domain waveform.

Such tools make graphical sound synthesis convenient. A history of graphical sound synthesis is beyond the scope of this article, but a few important figures include Evgeny Murzin, Percy Grainger, and Iannis Xenakis. In 1938, Evgeny Murzin invented a system to generate sound from a visible image; the design, based on the photo-optic sound technique used in cinematography, was implemented as the ANS synthesizer in 1958 (Kreichi 1995). Percy Grainger was also a pioneer with the "Free Music Machine" that he designed and built with Burnett Cross in 1952 (Lewis 1991); the device was able to generate sound from a drawing of an evolving pitch and amplitude. In 1977, Iannis Xenakis and associates built on these ideas when they created the famous UPIC (Unité Polyagogique Informatique du CEMAMu; Marino, Serra, and Raczinski 1993).

In this article, I explore the domain of graphical spectral analysis and synthesis in real-time situations. The technology has evolved so that now, not only can the phase vocoder perform analysis and synthesis in real time, but composers have access to a new conceptual approach: spectrum modifications considered as graphical processing. Nevertheless, the underlying matrix representation is still intimidating to many musicians. Consequently, the musical potential of this technique is as yet unfulfilled.

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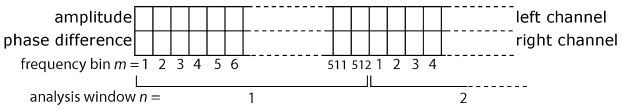
This article is intended as both a presentation of the potential of manipulating spectral sound data as matrices and a tutorial for musicians who want to implement such effects in the Max/MSP/Jitter environment. Throughout the article, I consider spectral analysis and synthesis as realized by the Fast Fourier Transform (FFT) and Inverse-FFT algorithms. I assume a familiarity with the FFT (Roads 1995) and the phase vocoder (Dolson 1986). To make the most of the examples, a familiarity with the Max/MSP environment is necessary, and a basic knowledge of the Jitter extension may be helpful.

I begin with a survey of the software currently available for working in this domain. I then show some improvements to the traditional phase vocoder used in both real time and *performance time.* (Whereas real-time treatments are applied on a live sound stream, performance-time treatments are transformations of sound files that are generated during a performance.) Finally, I present extensions to the popular real-time spectral processing method known as the *freeze*, to demonstrate that matrix processing can be useful in the context of real-time effects.

Spectral Sound Processing with Graphical Interaction

Several dedicated software products enable graphic rendering and/or editing of sounds through their sonogram. They generally do not work in real time, because a few years ago, real-time processing of complete spectral data was not possible on computers accessible to individual musicians. This calculation limitation led to the development of objects like IRCAM's Max/MSP external `iana~`, which reduces spectral data to a set of useful descriptors (Todoroff, Daubresse, and Fineberg 1995). After a quick survey of the current limitations of non-real-time software, we review the environments allowing FFT processing and visualization in real time.

Figure 1. FFT data recorded in a stereo buffer.



Non-Real-Time Tools

AudioSculpt, a program developed by IRCAM, is characterized by the high precision it offers as well as the possibility to customize advanced parameters for the FFT analysis (Bogaards, Röbel, and Rodet 2004). For instance, the user can adjust the analysis window size to a different value than the FFT size. Three automatic segmentation methods are provided and enable high-quality time stretching with transient preservation. Other important functions are frequency-bin independent dynamics processing (to be used for noise removal, for instance) and application of filters drawn on the sonogram. Advanced algorithms are available, including fundamental frequency estimation, calculation of spectral envelopes, and partial tracking. Synthesis may be realized in real time with sequenced values of parameters; these can, however, not be altered "on the fly." The application with graphical user interface runs on Mac OS X.

MetaSynth (Wenger and Spiegel 2005) applies graphical effects to FFT representations of sounds before resynthesis. The user can apply graphical filters to sonograms, including displacement maps, zooming, and blurring. A set of operations involving two sonograms is also available: blend, fade in/out, multiply, and bin-wise cross synthesis. It is also possible to use formant filters. The pitfall is that MetaSynth does not give control of the phase information given by the FFT: phases are randomized, and the user is given the choice among several modes of randomization. Although this might produce interesting sounds, it does not offer the most comprehensive range of possibilities for resynthesis.

There are too many other programs in this vein to list them all, and more are being developed each year. For instance, Spear (Klingbeil 2005) focuses on partial analysis, SoundHack (Polansky and Erbe 1996) has interesting algorithms for spectral mutation, and Praat (Boersma and Weenink 2006) is dedicated to speech processing. The latter two offer no interactive graphical control, though.

Real-Time Environments

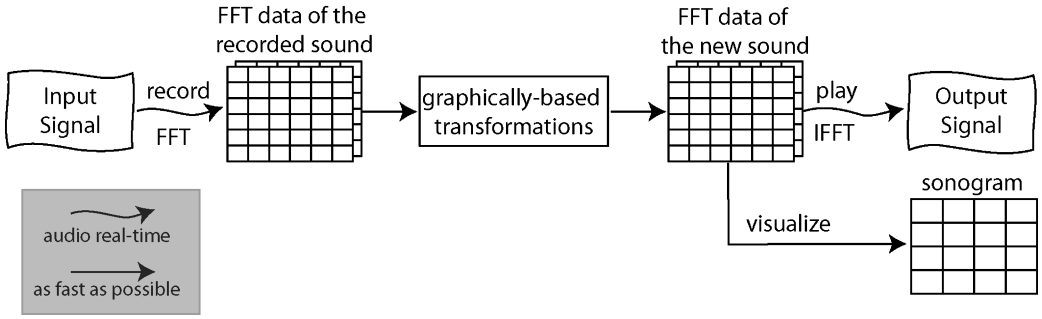
PureData and Max/MSP are two environments widely used by artists, composers, and researchers to process sound in real time. Both enable work in the spectral domain via FFT analysis/resynthesis. I focus on Max/MSP in this article, primarily because the MSP object `pfft~` simplifies the developer's task by clearly separating the time domain (outside the `pfft~` patcher) from the frequency domain (inside the patcher). When teaching, I find that the graphical border between the time and frequency domains facilitates the general understanding of spectral processing.

Using FFT inside Max/MSP may be seen as difficult because the FFT representation is two-dimensional (i.e., values belong to a time/frequency grid), whereas the audio stream and buffers are one-dimensional. Figure 1 shows how data can be stored when recording a spectrum into a stereo buffer.

The tension between the two-dimensional nature of the data and the one-dimensional framework makes coding of advanced processing patterns (e.g., visualization of a sonogram, evaluation of transients, and non-random modifications of spatial energy in the time/frequency grid) somewhat difficult. An example of treatment in this context is found in the work of Young and Lexer (2003), in which the energy in graphically selected frequency regions is mapped onto synthesis control parameters.

The lack of multidimensional data within Max/MSP led IRCAM to develop FTM (Schnell et al. 2005), an extension dedicated to multidimensional data and initially part of the jMax project. More specifically, the FTM `Gabor` objects (Schnell and Schwarz 2005) enable a "generalized granular synthesis," including spectral manipulations. FTM attempts to operate "Faster Than Music" in that the operations are not linked to the audio rate but

Figure 2. Phase vocoder design.



are done as fast as possible. FTM is intended to be cross-platform and is still under active development. However, the environment is not widely used at the moment, and it is not as easy to learn and use as Max/MSP.

In 2002, Cycling '74 introduced its own multidimensional data extension to Max/MSP: Jitter. This package is primarily used to generate graphics (including video and OpenGL rendering), but it enables more generally the manipulation of matrices inside Max/MSP. Like FTM, it performs operations on matrices as fast as possible. The seamless integration with Max/MSP makes implementation of audio/video links very simple (Jones and Nevile 2005).

Jitter is widely used by video artists. The learning curve for Max users is reasonable, thanks to the number of high-quality tutorials available. The ability to program in Javascript and Java within Max is also fully available to Jitter objects. Thus, I chose Jitter as a future-proof choice for implementing this project.

An Extended Phase Vocoder

Implementing a phase vocoder in Max/MSP/Jitter unlocks a variety of musical treatments that would remain impossible with more traditional approaches. Previous work using Jitter to process sound in the frequency domain include two sources: first, Luke Dubois's patch `jitter_pvoc_2D.pat`, distributed with Jitter, which shows a way to record FFT data in a matrix, to transform it (via zoom and rotation), and to play it back via the IFFT; second, an extension of this first work, where the authors introduce graphical transformations using a transfer matrix (Sedes, Courribet, and Thiebaut 2004). Both use a design similar to the one presented in Figure 2.

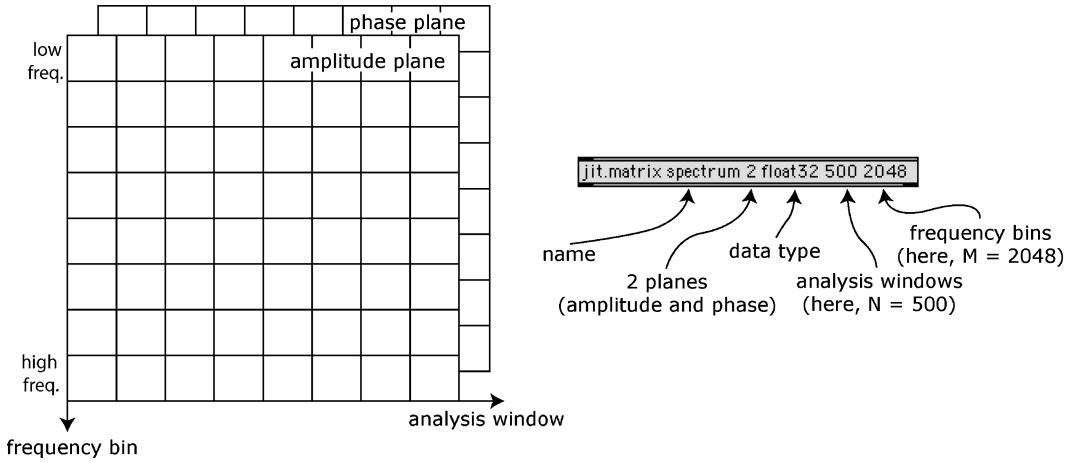
This section endeavors to make the phase-vocoder technique accessible, and it presents improvements in the resynthesis and introduces other approaches to graphically based transformations.

FFT Data in Matrices

An FFT yields Cartesian values, usually translated to their equivalent polar representation. Working with Cartesian coordinates is sometimes useful for minimizing CPU usage. However, to easily achieve the widest and most perceptively meaningful range of transformations before resynthesis, we will work with magnitudes and phase differences (i.e., the bin-by-bin differences between phase values in adjacent frames).

Figure 3 shows how I store FFT data in a Jitter matrix throughout this article. We use 32-bit floating-point numbers here, the same data type that MSP uses. The grid height is the number of frequency bins, namely, half the FFT size, and its length is the number of analysis windows (also called frames in the FFT literature). It is logical to use a two-plane matrix: the first is used for magnitudes, the second for phase differences.

Figure 3. FFT representation of a sound stored in a matrix.



Interacting with a Sonogram

The sonogram represents the distribution of energy at different frequencies over time. To interact with this representation of the spectrum, we must scale the dimension of the picture to the dimension of the matrix holding the FFT data. To map the matrix cell coordinates to frequency and time domains, we use the following relations. First, time position t (in seconds) is given by

$$t = n \times \frac{WindowSize}{OverlapFactor} \times \frac{1}{sr} \quad (1)$$

where n is the number of frames, sr is the sampling rate (Hz), and *WindowSize* is given in samples. Second, the center frequency f_c (Hz) of the frequency bin m is

$$f_c = m \times \frac{sr}{FFTSize} \quad (2)$$

Third, assuming no more than one frequency is present in each frequency bin in the analyzed signal, its value in Hz can be expressed as

$$f = f_c + \Delta\phi \times \frac{sr}{2\pi \times WindowSize} \quad (3)$$

where $\Delta\phi$ is the phase difference, wrapped within the range $[-\pi, \pi]$ (Moore 1990). Note that with the `pfft~` object, the window size (i.e., frame size) is the same as the FFT size. (In other words, there is no zero-padding.)

The patch in Figure 4 shows a simple architecture for interacting with the sonogram. Equations 1 and 2 and part of Equation 3 are implemented inside `patcher` (or `p`) objects. To reduce computational cost, the amplitude plane of the matrix holding the FFT data is converted to a low-resolution matrix before the display adjustments (i.e., the inversion of low/high frequencies and inversion of black and white, because by default, 0 corresponds to black and 1 to white in Jitter). The OpenGL implementation of the display is more efficient and flexible, but it is beyond the scope of this article.

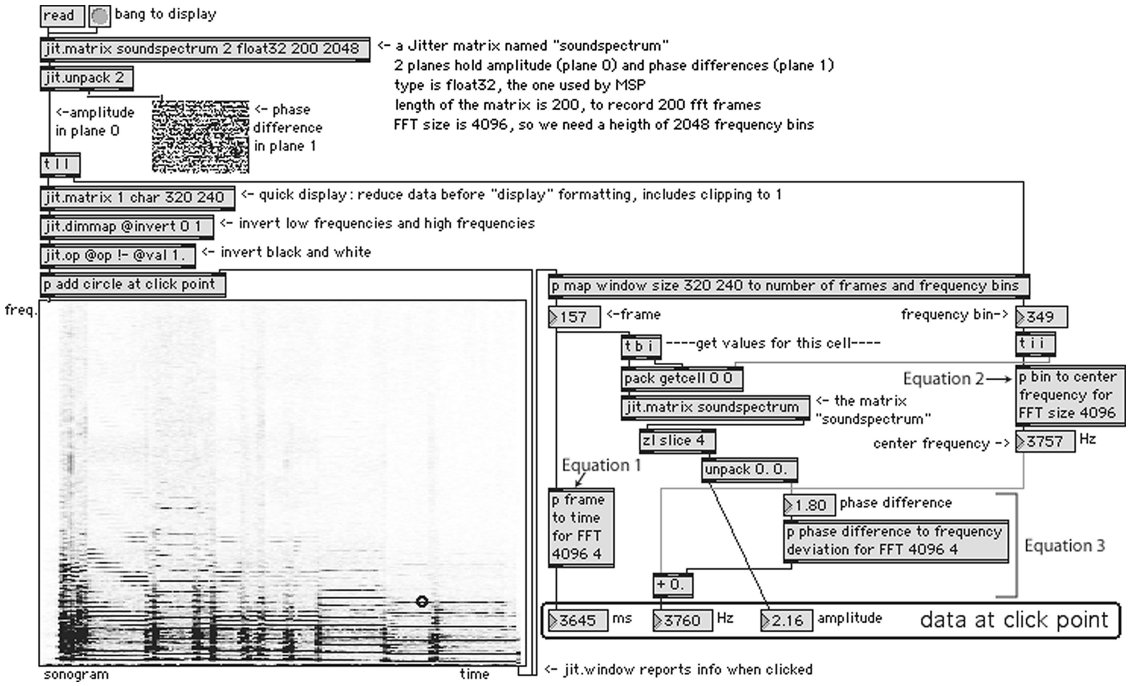
Recording

When implementing the recording of FFT data into a matrix, we must synchronize the frame iteration with the FFT bin index. The frame iteration must happen precisely when the bin index leaps back to 0 at the beginning of a new analysis window. Luke Dubois's patch `jitter_pvoc_2D.pat` contains one implementation of a solution. (See the object `count~` in `[pfft~ jit.recordfft~.pat]`.)

Playback

Figure 5 shows a simple playback algorithm using an IFFT. A control signal sets the current frame to read.

Figure 4. Interaction with a sonogram.



To play back the sound at the original speed, the control signal must be incremented by one frame at a period equal to the hop size. Hence, the frequency f (in Hz) of the `phasor~` object driving the frame number is

$$f = \frac{sr}{N \times HopSize} \times PlaybackRate \qquad (4)$$

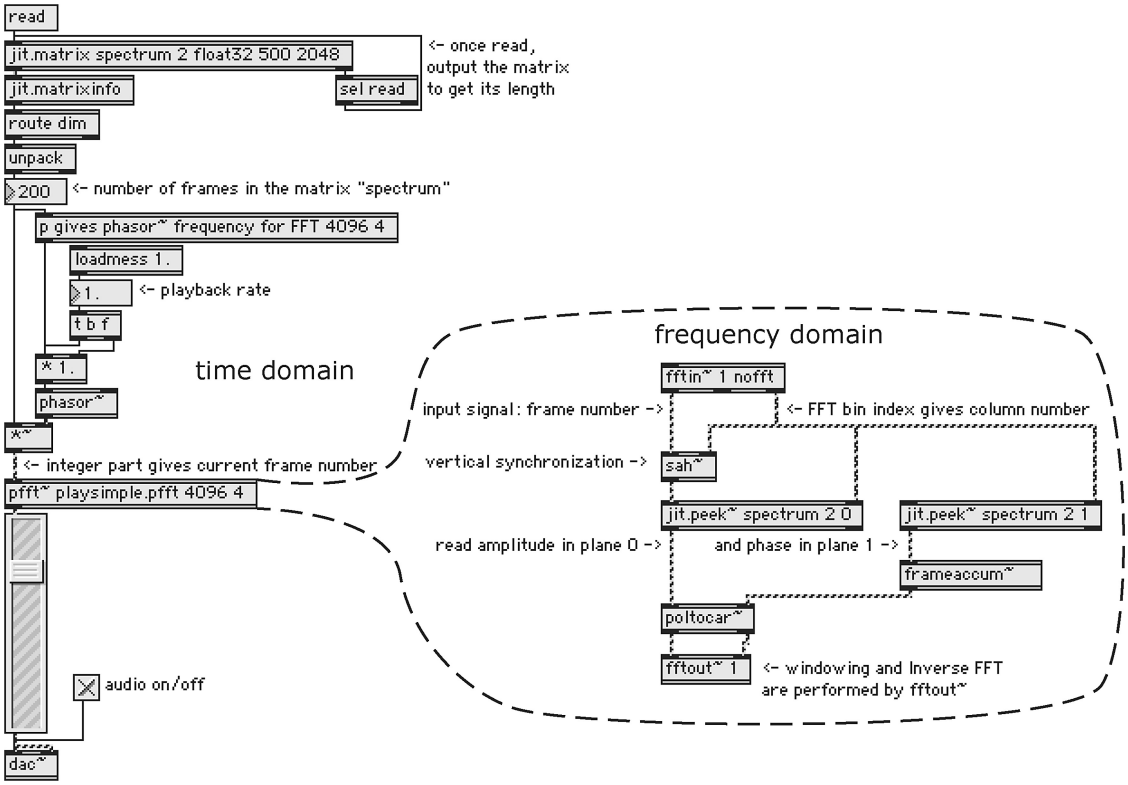
where N is the total number of analysis windows. Here, *PlaybackRate* sets the apparent speed of the sound: 1 is normal, 2 is twice as fast, 0 freezes one frame, and –1 plays the sound backwards.

Advanced Playback

In extreme time stretching, the two main artifacts of the phase vocoder are the "frame effect" and the time-stretching of transients. In my composition *Plex* for solo instrument and live electronics, ten seconds are recorded at the beginning of the piece and played back 36 times more slowly, spread over 6 minutes. Considering an FFT with a hop size of 1,024 samples and a sampling rate of 44,100 Hz, each analysis window with an original duration of 23 msec is stretched out to 836 msec. During synthesis with the traditional phase-vocoder method, the leap from one frame to the following one is audible with a low playback speed. This particular artifact, the "frame effect," has not received much attention in the literature. I will describe two methods to interpolate spectra between two recorded FFT frames, one of which can easily be extended to a real-time "blurring" effect.

When humans slow down the pacing of a spoken sentence, they are bound to slow down vowels more than consonants, some of which are physically impossible to slow down. Similarly, musicians can naturally slow down the tempo of a phrase without slowing down the attacks. That is why time-stretching a sound uniformly, including transients, does not sound natural. Transient preservation in the phase vocoder has been studied, and several efficient algorithms have been presented (Laroche and Dolson 1999; Röbel 2003). Nevertheless, these propositions have not been created in a form adapted to real-time environments. I present a simple, more straightforward approach that computes a transient value for

Figure 5. A simple player.



each frame and uses it during resynthesis. That will naturally lead to an algorithm for segmentation, designed for performance time as well.

Removing the "Frame Effect"

This section shows two algorithms interpolating intermediary spectra between two "regular" FFT frames, thus smoothing transitions and diminishing the frame effect. In both cases, a floating-point number describes the frame to be resynthesized: for instance, 4.5 for an interpolation halfway between frames 4 and 5.

In the first method, the synthesis module continuously plays a one-frame spectrum stored in a one-column matrix. As illustrated in Figure 6, this matrix is fed with linear interpolations between two frames from the recorded spectrum, computed with the `jit.xfade` object. Values from both amplitude and phase difference planes are interpolated. With a frame number of 4.6, the value in a cell of the interpolated matrix is the sum of 40 percent of the value in the corresponding cell in frame 4 and 60 percent of the value in frame 5.

My second approach is a controlled stochastic spectral synthesis. For each frame to be synthesized, each amplitude/phase-difference pair is copied from the recorded FFT data, either from the corresponding frame or from the next frame. The probability of picking up values in the next frame instead of the current frame is the fractional value given by the user. For instance, a frame number of 4.6 results in a probability of 0.4 that values are copied from the original frame 4 and a probability of 0.6 that values are copied from the original frame 5. This random choice is made for each frequency bin within each synthesis frame, such that two successive synthesized frames do not have the same spectral content, even if the specification is the same. The implementation shown in Figure 7 shows a small

Figure 6. Interpolated spectrum between two analysis windows.

Figure 7. A smooth player.

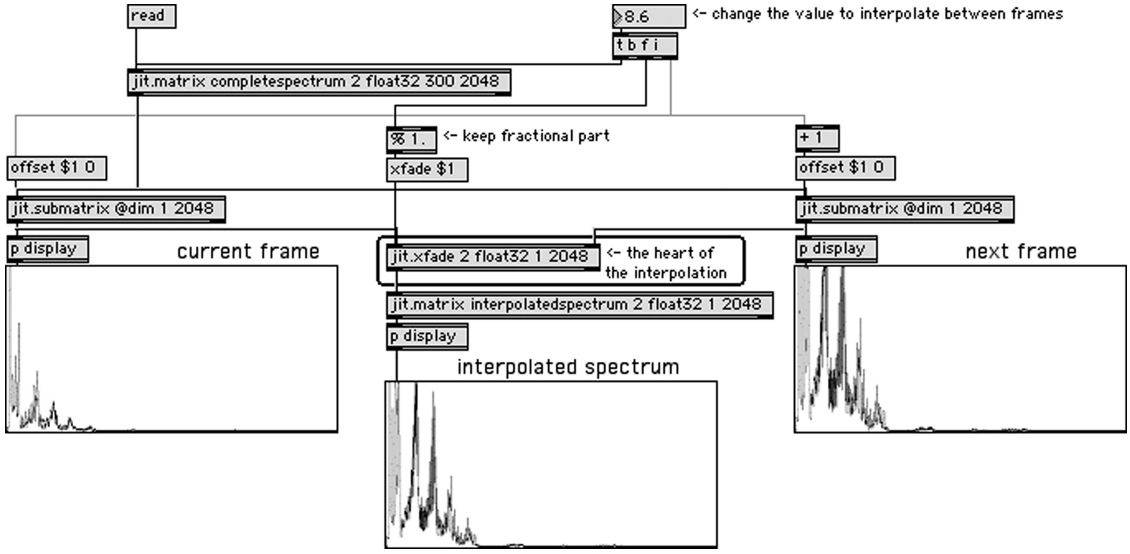


Figure 6

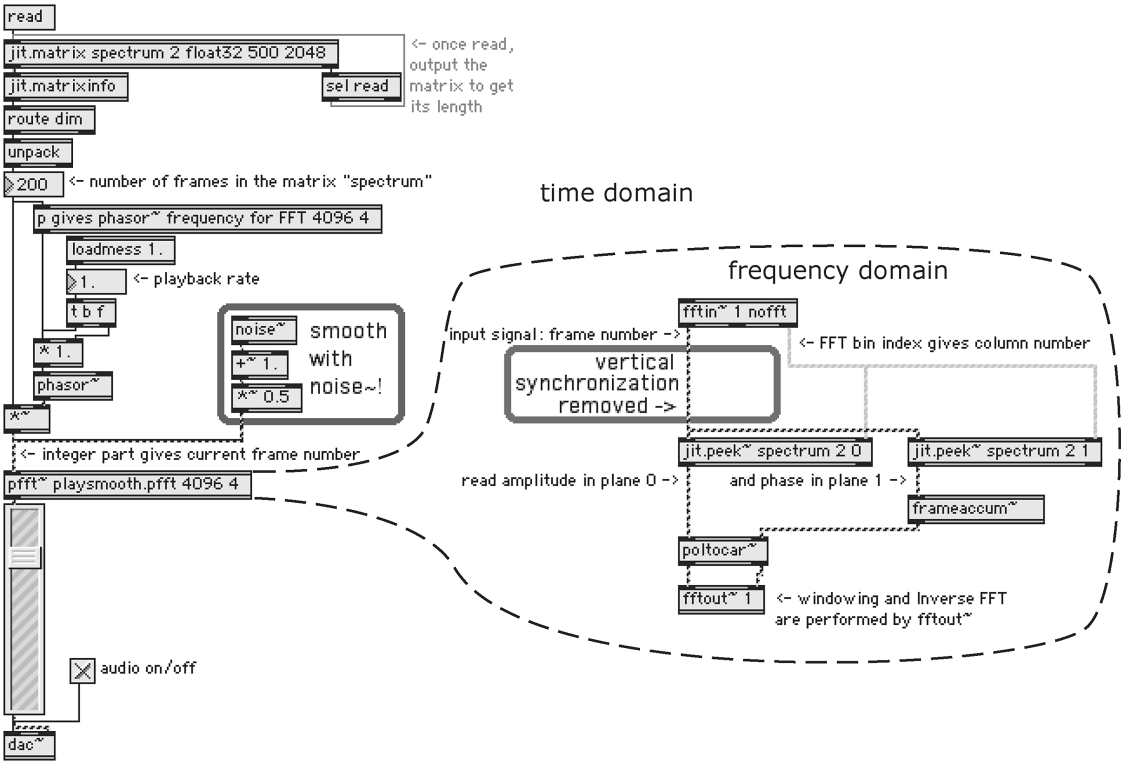
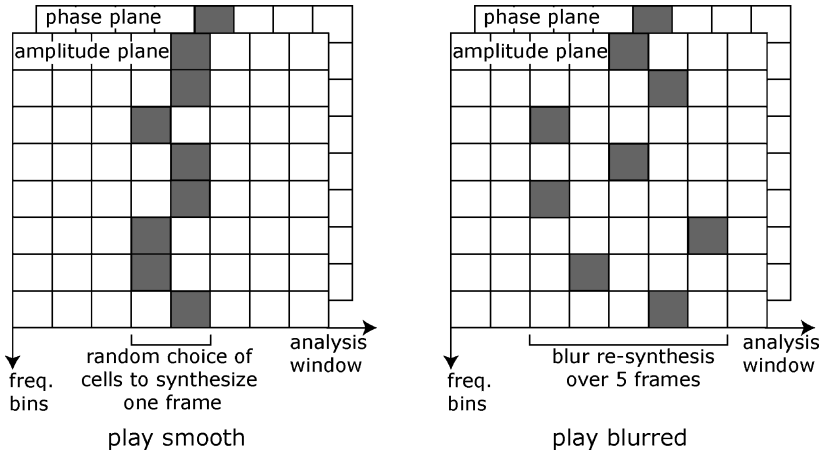


Figure 7

Figure 8. Stochastic synthesis between two frames, blur over several frames.



modification to the engine presented Figure 5. A signal-rate noise with amplitude from 0 to 1 is added to the frame number. While reading the FFT data matrix, `jit.peek~` truncates the floating-point value to an integer giving the frame number. The vertical synchronization is removed to enable the frame number to change at any time. Increasing the floating-point value in the control signal creates an increased probability of reading the next frame.

This method can be readily extrapolated to blur any number of frames over time with negligible additional cost in CPU power. Instead of adding a noise between 0 and 1, we scale it to [0, R], where R is the blur width in frames. If C is the index of the current frame, each pair of values is then copied from a frame randomly chosen between frames C and $C + R$, as shown in Figure 8. The musical result is an ever-moving blur of the sound, improving the quality of frozen sounds in many cases. The blur width can be controlled at the audio rate.

Both the interpolated frame and the stochastic playback methods produce high-quality results during extremely slow playback, but they are useful in different contexts. The interpolated frames often sound more natural. Because it lacks vertical coherence, the stochastic method may sound less natural, but it presents other advantages. First, it is completely implemented in the "perform" thread of Max/MSP; as a result, CPU use is constant regardless of the parameters, including playback rate. However, the interpolated spectrum must be computed in the low-priority thread, meaning that the CPU load increases with the number of frames to calculate. Second, the stochastic method offers a built-in blurring effect, whereas achieving this effect with matrix interpolation would require additional programming and be less responsive. In what follows, I continue to develop this second method, because it adheres to the original phase-vocoder architecture and is more flexible.

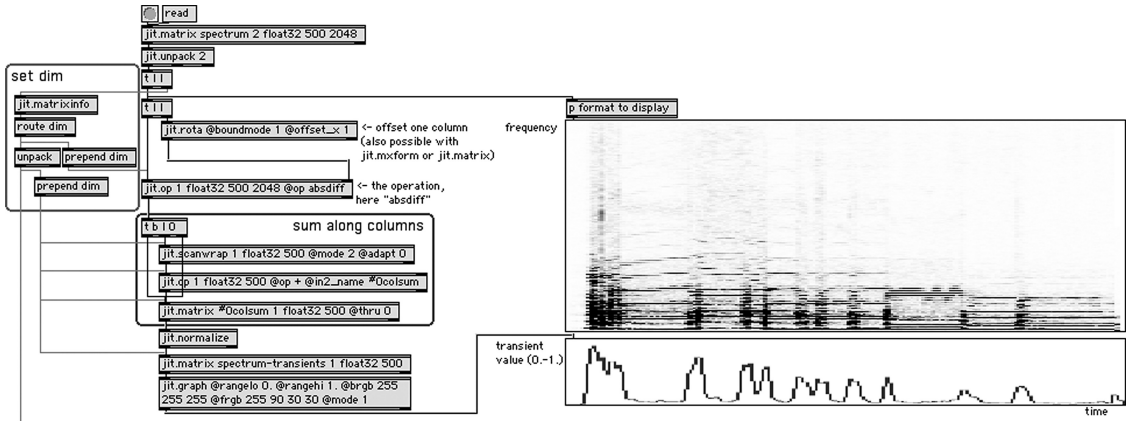
Transient Preservation

I propose a simple algorithm for performance-time transient evaluation, along with a complementary algorithm to play a sound at a speed proportional to the transient value. My approach consists in assigning to each frame a transient value, defined as the spectral distance between this frame and the preceding frame, normalized to [0, 1] over the set of frames.

Several choices are possible for the measure of the spectral distance between two frames. The Euclidean distance is the most obvious choice. Given M frequency bins, with $a_{m,n}$ the amplitude in bin m of frame n, the Euclidean distance between frames n and $n-1$ is

$$t_n = \sqrt{\sum_{m=0}^{M-1} (a_{m,n} - a_{m,n-1})^2} \qquad (5)$$

Figure 9. A transient value for each frame.



In this basic formula, we can modify two parameters that influence quality and efficiency: first, the set of descriptors out of which we calculate the distance, and second, the distance measure itself. The set of descriptors used in Equation 5 is the complete set of frequency bins: applying Euclidean distance weights high frequencies as much as low ones, which is not perceptually accurate. For a result closer to human perception, we would weight the different frequencies according to the ear's physiology, for instance by using a logarithmic scale for the amplitudes and by mapping the frequency bins non-linearly to a set of relevant descriptors like the 24 Bark or Mel-frequency coefficients. In terms of CPU load, distance is easier to calculate over 24 coefficients than over the whole range of frequency bins, but the calculation of the coefficients makes the overall process more computationally expensive. Although using the Euclidean distance makes mathematical sense, the square root is a relatively expensive calculation. To reduce computation load, in Figure 9 I use in a rough approximation of the Euclidean distance:

$$t_n = \sum_{m=0}^{M-1} \left| a_{m,n} - a_{m,n-1} \right| \qquad (6)$$

The playback part of the vocoder can use the transient value to drive the playback rate. In the patch presented Figure 10, the user specifies the playback rate as a transient rate r_{trans} (the playback rate for the greatest transient value) and a stationary rate r_{stat} (the playback rate for the most stationary part of the sound). Given the current frame's transient value tr, the instantaneous playback rate r_{inst} is given by

$$r_{inst} = r_{stat} + tr \times (r_{trans} - r_{stat}) \qquad (7)$$

The musical result of a continuous transient value between 0 and 1 is interesting, because it offers a built-in protection against errors of classification as transient or stationary. This is especially useful in a performance-time context. A binary transient value could, of course, be readily implemented by a comparison to a threshold applied to all the values of the transient matrix with a `jit.op` object.

Controlling the blur width (see Figure 8) with the transient value may be musically useful as well. The most basic idea is to make the blur width inversely proportional to the transient value, as presented in Figure 11. Similarly to the playback rate, the instantaneous blur width b_{inst} is given by

$$b_{inst} = b_{stat} + tr \times (b_{trans} - b_{stat}) \qquad (8)$$

where tr is the current frame transient value, b_{stat} is the stationary blur width, and $b_{trans}$is the transient blur width.

Segmentation

This section describes a performance-time method of segmentation based on transient values of successive frames. I place a marker where the spectral changes

Figure 10. Re-synthesizing at self-adapting speed.

Figure 11. Blur width controlled by transient value. During the playback of stationary parts of the sound (left), the blurred region is wider than during the playback of transients (right).

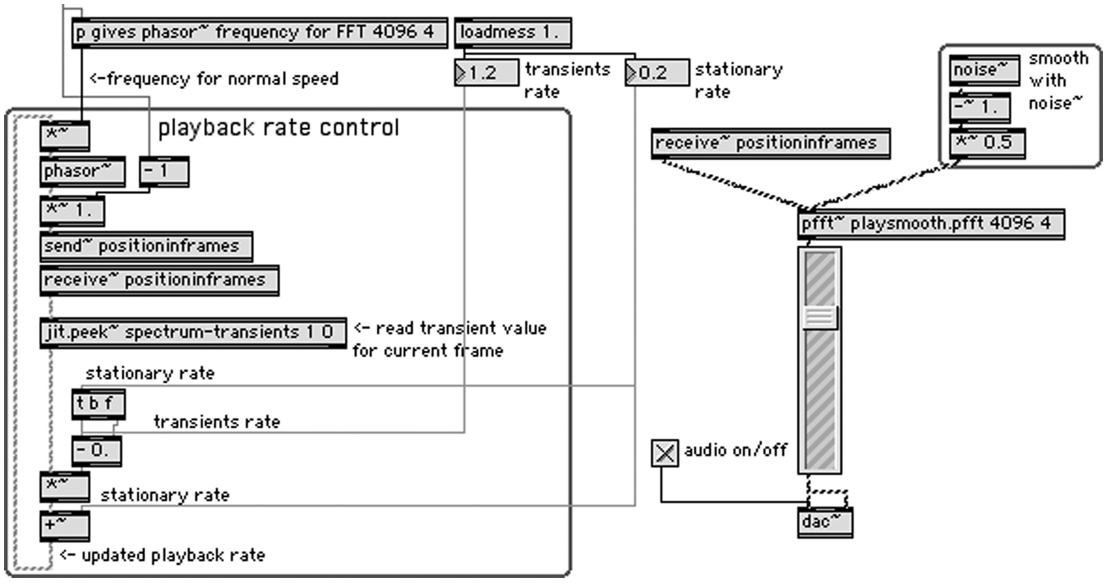


Figure 10

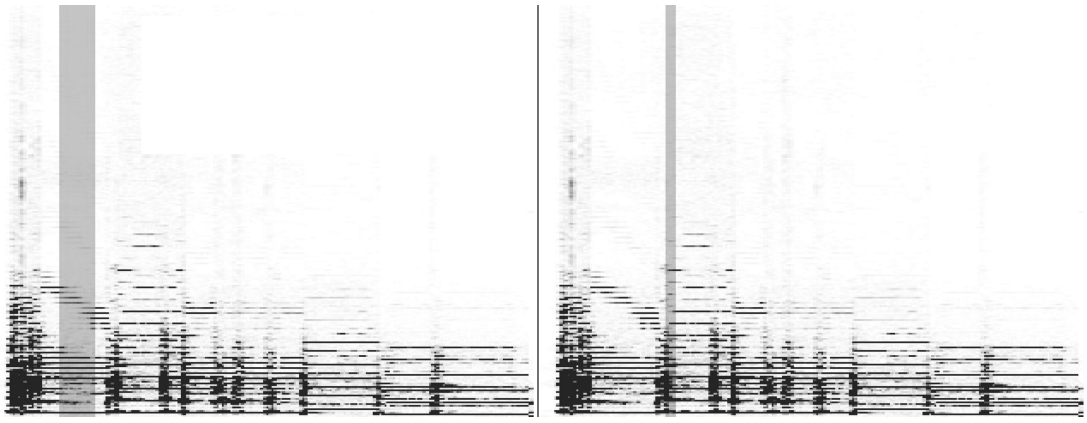


Figure 11

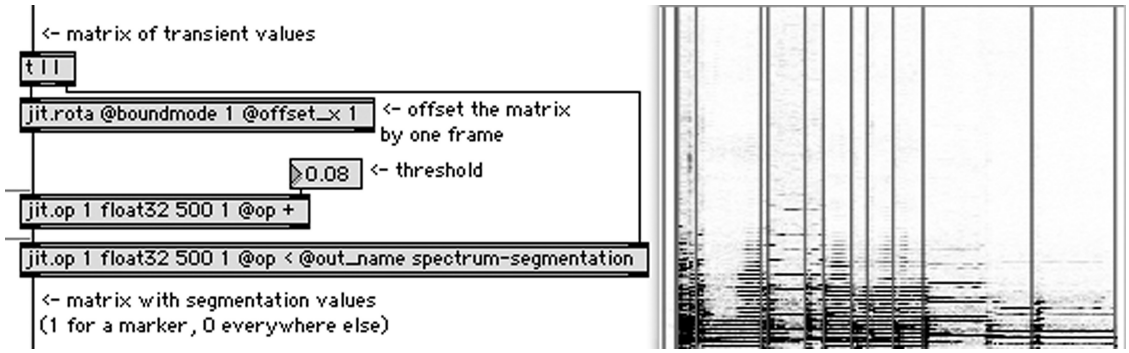
from frame to frame are greater than a given threshold. As Figure 12 shows, frame n is marked if and only if its transient value is greater than or equal to the transient value in the preceding frame plus a threshold h, that is, whenever

$$t_n \geq t_{n-1} + h \tag{9}$$

The formula used to calculate transient values appears to have a great influence on the segmentation result. In the preceding section, we used Euclidean distance or an approximation by absolute difference (Equations 5 and 6). Both expressions yield a useful auto-adaptive playback rate, but my experience is that the following expression is preferable for ulterior segmentation with Equation 9:

$$t_n = \sum_{m=0}^{M-1} \frac{a_{m,n}}{a_{m,n-1}} \tag{10}$$

Figure 12. Automatic performance-time segmentation.



Indeed, whereas Euclidean distance or absolute difference give spectrum changes a lower weight at locally low amplitudes than at high amplitudes, the division in Equation 10 enables a scaling of the degree of spectrum modification to the local level of amplitude. Musical applications include real-time leaps from one segmentation marker to another, resulting in a meaningful shuffling of the sound.

An important point to retain from this section is that, whereas ideas may materialize first as iterative expressions (e.g., see the summations in Equations 5, 6, and 10), the implementation in Jitter is reduced to a small set of operations on matrices. To take full advantage of Jitter, we implement parallel equivalents to iterative formulas.

Graphically Based Transformations

Now, we explore several ways of transforming a sound through its FFT representation stored in two-dimensional matrices. The four categories I examine are direct transformations, use of masks, interaction between sounds, and "mosaicing."

Direct Transformations

The most straightforward transformation consists of the application of a matrix operation to an FFT data matrix. In Jitter, we can use all the objects that work with 32-bit floating-point numbers. A flexible implementation of such direct transformations is possible thanks to `jit.expr`, an object that parses and evaluates expressions to an output matrix. Moreover, all Jitter operators compatible with 32-bit floating-point numbers are available in `jit.expr`.

Phase information is important for the quality of resynthesized sounds. The choice to apply a treatment to amplitude and phase difference or only to amplitude, or to apply a different treatment to the phase-difference plane, is easily implemented whether in `jit.expr` or with `jit.pack` and `jit.unpack` objects. This choice depends on the particular situation and must generally be made after experimenting.

In Figure 13, the expression attribute of the second `jit.expr` object is `[''gtp(in[0].p[0]\, 1.3)'' ''in[0].p[1]'']`. The first part of the expression is evaluated onto plane 0 of the output matrix; it yields the amplitude of the new sound. It applies the operator `gtp` (which passes its argument if its value is greater than the threshold, otherwise, it yields 0) to the amplitude plane of the incoming matrix with the threshold 1.3. It is a rough denoiser. (In the previous expression, `in[0].p[0]` is plane 0 of the matrix in input 0.) The second part of the expression produces on plane one the first plane of the incoming matrix (i.e., `in[0].p[1]`); the phase is kept unchanged.

Masks

Masks enable the gradual application of an effect to a sound, thus refining the sound quality. A different percentage of the effect can be used in every frequency bin and every frame. In this article,

Figure 13. Graphical transformations.

Figure 14. Two designs to use masks.

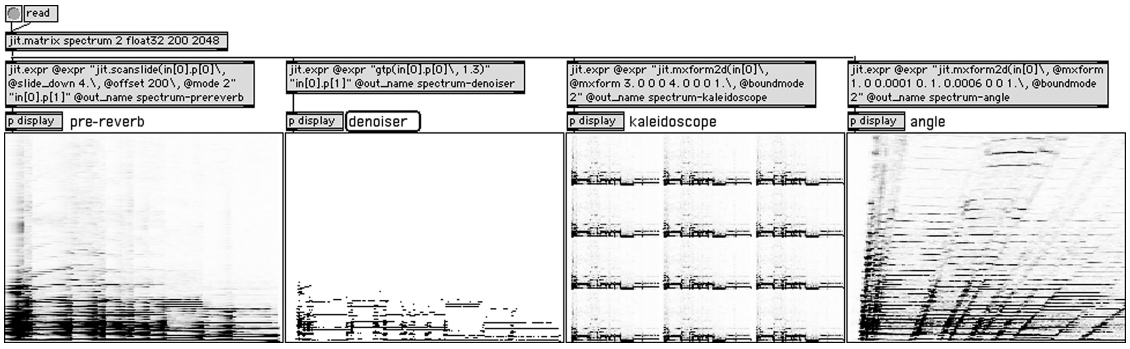


Figure 13

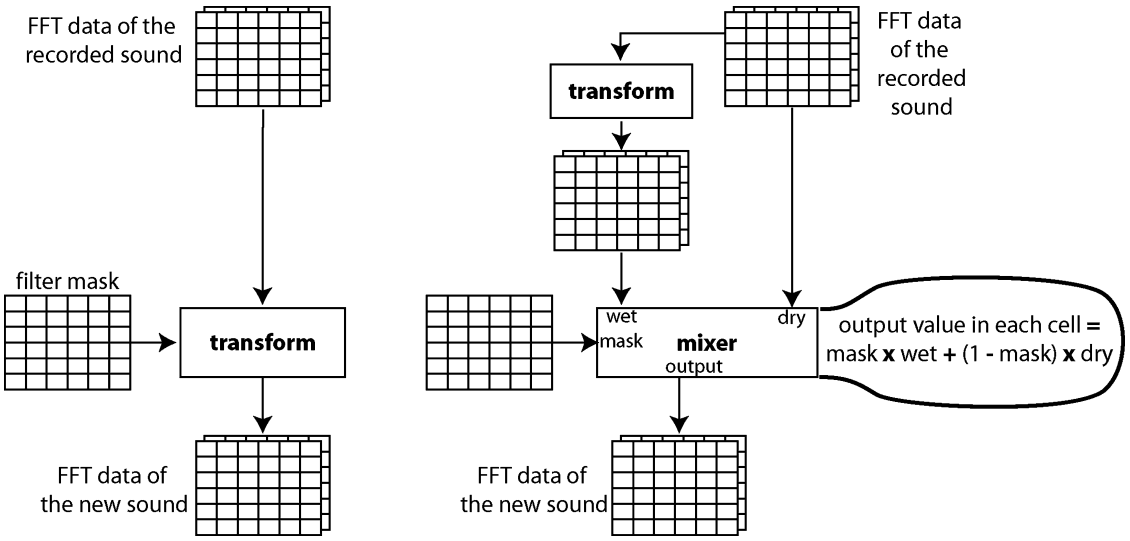


Figure 14

I limit the definition of a mask to a grid of the same resolution as the FFT analysis of the sound, with values between 0 and 1. In Jitter, that means a one-dimensional, 32-bit floating-point matrix, of the same dimension as the sound matrix.

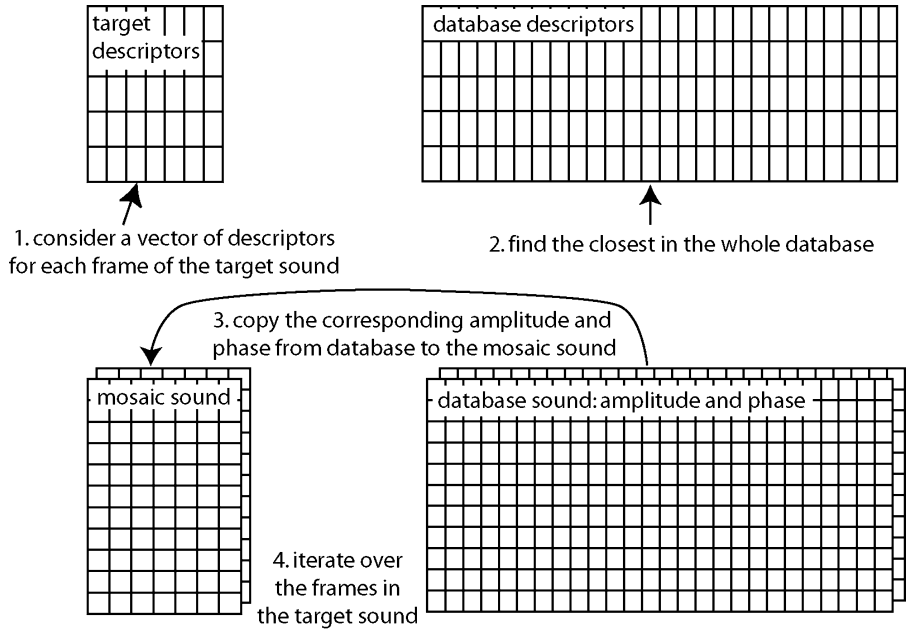
A mask can be arbitrarily generated, interpolated from a graphical file, drawn by the user on top of the sound sonogram, or computed from another sound or the sound itself. For example, let us consider a mask made of the amplitudes (or the spectral envelope) of one sound, mapped to [0, 1]. Multiplying the amplitudes of a source sound with this mask is equivalent to applying a source-filter cross synthesis.

Figure 14 presents two possible designs. In the left version, the processing uses the mask to produce the final result; this is memory-efficient. In the right version, the original sound is mixed with the processed one; this allows a responsive control on the mix that can be adjusted without re-calculation of the sound transformation.

Interaction of Sounds

Generalized cross synthesis needs both amplitude and phase difference from both sounds. Similarly,

Figure 15. Context-free mosaicing algorithm.



interpolating and morphing sounds typically requires amplitude and phase information from both sounds.

It is easy to implement interpolation between sounds with `jit.xfade`, in the same way as interpolation between frames in Figure 6. An interesting musical effect results from blurring the cross-faded sound with filters such as `jit.streak`, `jit.sprinkle` , and `jit.plur`.

As opposed to an interpolated sound, a morphing between two sounds evolves from the first to the second within the duration of the synthesized sound itself. Such a morphing can be implemented as an interpolation combined with three masks: one for the disappearance of the first source sound, a second for the appearance of the second sound, and a third one to apply a variable amount of blurring.

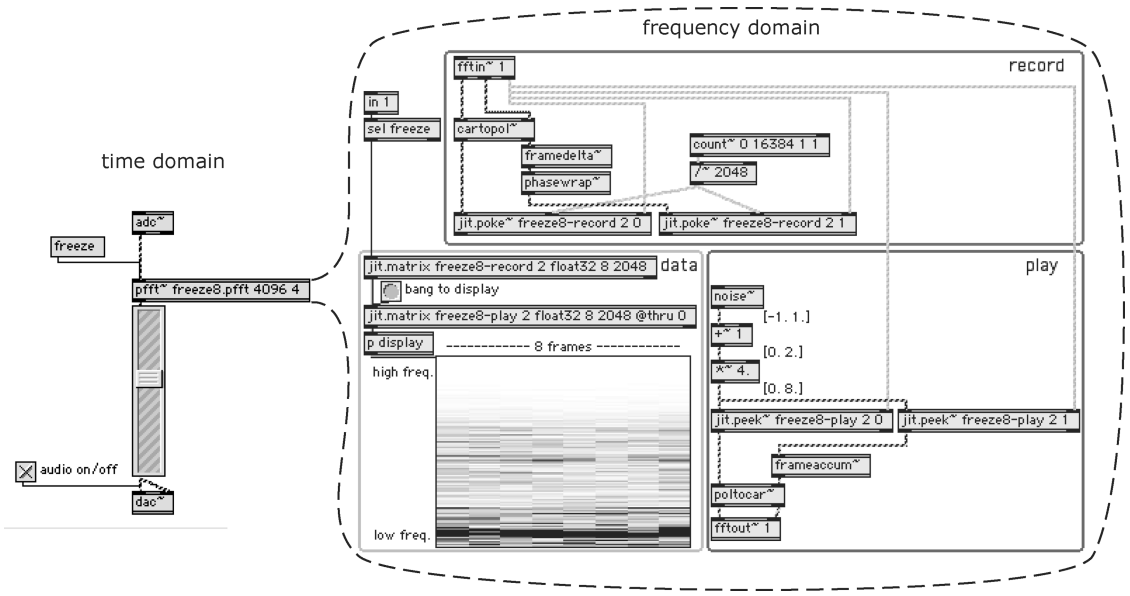
Mosaicing

Mosaicing is a subset of concatenative synthesis, where a target sound is imitated as a sequence of best-matching elements from a database (Schwarz 2006). In the context of the phase vocoder, I provide a simple mosaicing by spectral frame similarity, similar to the Spectral Reanimation technique (Lyon 2003). Before resynthesis, each FFT frame in the target sound is replaced by the closest frame in the database irrespective of the context. The algorithm is illustrated in Figure 15.

Three factors influence the speed and quality of the resulting sound. First, the amplitude scale might be linear (less CPU intensive) or logarithmic (closer to perception). Second, the descriptors for each frame can take variable sizes. The set of amplitudes in each frequency bin is directly available, but it is a large vector. A linear interpolation to a reduced number of bands, a set of analyzed features (pitch, centroid, etc.), and a nonlinear interpolation to the 24 Bark or Mel-frequency coefficients are other possible vectors. The third factor is the distance used to find the closest vector (Euclidean distance, or an approximation sparing the square-root calculation).

My choice has been to use a logarithmic amplitude scale, a vector of Bark coefficients, and Euclidean distance or an approximation similar to Equation 6. The Bark coefficients, reflecting ear physiology, are well suited to this task.

Figure 16. Real-time stochastic freeze implementation.



Real-Time Freeze and Beyond

The matrix approach to spectral treatments allows not only a variety of performance-time extensions to the phase vocoder but also improvements in real-time processing, such as freezing a sound and transforming a melody into a chord.

Real-Time Freeze

A simple way to freeze a sound in real time is to resynthesize one spectral frame continuously. I improve the sound quality by freezing several frames at once and then resynthesizing the complete set of frames with the stochastic blurring technique described previously (Figure 8).

My implementation features a continuous spectral recording of the input sound into a circular buffer of eight frames (an eight-column matrix). This matrix is copied into the playback matrix upon receipt of the `freeze` message.

Automatic Cross-Fade

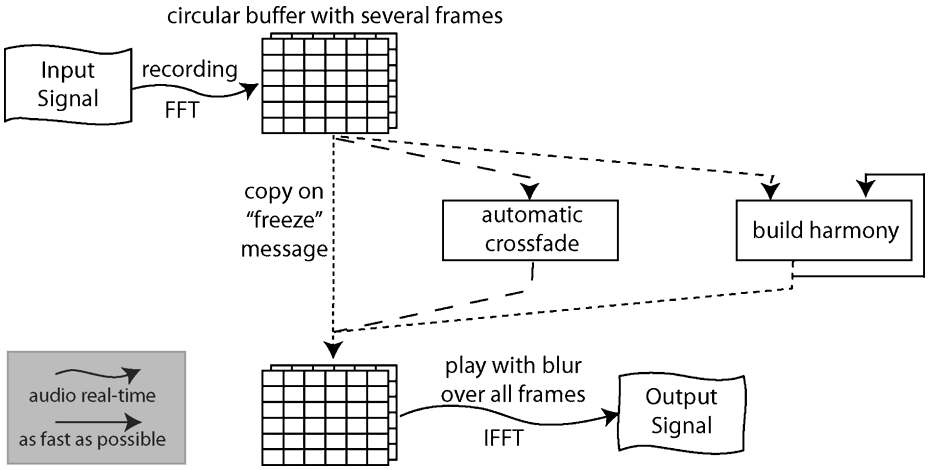
The stochastic freeze presented in Figure 16 switches abruptly from one frozen sound to the next. A transition sub-patch between the matrix to freeze (`freeze8-record`) and the matrix currently playing (`freeze8-play`) can be used to cross-fade when a new sound is frozen. Such sub-patches incorporate first a counter to generate a number of transition matrices, and second a cross-fading object such as `jit.xfade` controlled by a user-drawn graph specifying the cross-fade curve.

Melody to Harmony

In my composition *Aqua* for solo double bass, aquaphon, clarinet, and live electronics, arpeggios are transformed in real time into chords with a modified freeze tool. As shown in Figure 17, the currently synthesized spectrum is added to the incoming one via a feedback loop. The "build harmony" block uses the current output synthesis data and the new window analysis to calculate a matrix reflecting the addition of the new note. The first solution for this computation is to average the energy in each frequency bin over all frozen spectra:

$$a_n = \frac{\sum_{i=1}^{n} a_i}{n} \tag{11}$$

Figure 17. Real-time freeze design with extensions.



where n is the number of frames to average, a_n is the average amplitude, and a_i is the amplitude in frame i. The same formula can be used for phase differences. The implementation requires a recursive equivalent to Equation 11:

$$a_{n+1} = \frac{n \times a_n + a_{i+1}}{n+1} \tag{12}$$

Note that in the case of stochastic blurred resynthesis over eight frames, this operation is done independently for each frame.

However, the solution I ended up using in concert is given by

$$a_n = \frac{\sum_{i=1}^{n} a_i}{\sqrt{n}} \tag{13}$$

which is written and implemented recursively as

$$a_{n+1} = \frac{\sqrt{n} \times a_n + a_{i+1}}{\sqrt{n+1}} \tag{14}$$

Indeed, the latter formula produces a more powerful sound, compensating for the low-amplitude frames that may be recorded.

Conclusion

This article provides an overview of techniques to perform spectral audio treatments using matrices in the environment Max/MSP/Jitter. I have presented a paradigm for extensions of the phase vocoder in which advanced graphical processing is possible in performance time. The "composition of sound treatments" described by Hans Tutschku (Nez 2003) is now available not only in the studio, but also during a concert. I have also described an improvement to the real-time effect known as *freeze*. I hope that these musical tools will help create not only new sounds, but also new compositional approaches. Some patches described in this article, complementary patches, and sound examples are available online at www.jeanfrancoischarles.com.

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