ABSTRACT

Title of dissertation: BIT OF NOSTALGIA… FOR ONE OR TWO PERCUSSIONISTS AND LIVE ELECTRONICS PERFORMER

Michael Boyd, Doctor of Musical Arts, 2006

Dissertation directed by: Professor Thomas DeLio
School of Music

Bit of nostalgia… is a work for one or two percussionists and a live electronics performer that explores performer creativity through a graphic score and investigates the ways that the objects performers interact with (instruments) shape their actions/performances. The percussionist(s) take an active role in designing the stage setup for each performance by superimposing a grid on the performance space, and filling at least half of the sectors with combinations of instrument-types listed in the piece’s instructions (including objects made of metal, wood, glass, paper, plastic, and stone). Each sector that contains groups of instruments also contains a music stand holding three of eighteen closely related score pages. The similarities inherent in the various score pages requires that performers frequently reinterpret quasi-redundant visual materials with greatly varying groups of instruments (and objects), emphasizing the differences between each instrument group.
While the performance proceeds, another performer interprets the same score using Cycling 74’s MAX/MSP (software that accomplishes real-time sound synthesis and processing) to process and playback sound segments from recordings of previous rehearsals and/or performances. To accomplish this, the computer performer utilizes some or all of nine MAX/MSP patches (single windows containing a user designed graphical interface) of my design that incorporate differently controlled ring modulation, filtering, and delay in isolation or various combinations. The percussionists directly respond to these sounds as well as each other while interpreting certain pages of the score that contain the letters I, O, and T (signifying imitate, oppose, and transform respectively). These letters direct the performers to address sounds/actions produced by the other performer, themselves, or the electronics through their interpretations. Through these interactions, I hope to bring a sense of self-history into the piece and create an interesting notion of depth which reflects a broader perspective of what constitutes a “work.” Whereas one typically thinks of an artwork as a fixed entity such as a score, I am trying to overtly link and interconnect otherwise marginalized and disparate aspects that contribute to the totality of this piece such as rehearsals and performances.
BIT OF NOSTALGIA… FOR ONE OR TWO PERCUSSIONISTS AND LIVE ELECTRONICS PERFORMER

by

Michael Boyd

Dissertation submitted to the Faculty of the Graduate School of the University of Maryland, College Park in partial fulfillment of the requirements for the degree of Doctor of Musical Arts

2006

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Performance Instructions

This piece is for one or two percussionists and a second/third performer manipulating live electronics.

Performance Space Segmentation

The performance space is conceptually divided into a number of equal-sized sectors – nine for a duo (single percussionist) performance (3x3, see Grid 1), and sixteen for a trio (4x4, see Grid 2). Each sector will either remain empty or contain a music stand and instrument(s); each music stand should hold one packet of three score pages (some performances may not use all score packets depending on how many sectors are filled). At least two, but no more than half of the sectors should be empty.

Grid 1 (duo)

<p>| | | |</p>
<table>
<thead>
<tr>
<th></th>
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<th></th>
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<tbody>
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</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
The remaining sectors (those not left empty) can be filled by using the list of instrument configurations provided (see the chart below); note that all non-empty sectors should be adjacent with at least one other non-empty sector (connection by corner is acceptable). The instrumental configurations are arranged into categories, and at least one configuration from each should be used in every performance. The configuration designations simply indicate the material that comprises the majority of each instrument (metal, wood, plastic, paper, stone, or glass); these instruments can be traditional percussion instruments, non-percussion instruments, or objects not typically associated with musical performance (some of each type should be used). The actual arrangement of instruments/objects in each sector is left to the performer’s discretion, though the score should be visible when interacting with all instruments. The electronics performer should be positioned with on the periphery of or just beyond the performance space with a computer, mixer, and so forth. Two speakers should be interspersed in the space at the performers’ discretion.

### Instrument configurations

<table>
<thead>
<tr>
<th>Category 1</th>
<th>Category 2</th>
<th>Category 3</th>
<th>Category 4</th>
<th>Category 5</th>
</tr>
</thead>
<tbody>
<tr>
<td>Music stand alone (no other objects)</td>
<td>Multiple metal</td>
<td>1 wood</td>
<td>1 glass, 1 stone</td>
<td>Multiple metal, plastic, paper</td>
</tr>
<tr>
<td></td>
<td>Multiple plastic</td>
<td>1 paper</td>
<td>1 wood, 1 paper</td>
<td>Multiple stone, wood, plastic</td>
</tr>
<tr>
<td></td>
<td>Multiple glass</td>
<td>1 stone</td>
<td>1 metal, 1 glass</td>
<td>Multiple paper, glass, plastic</td>
</tr>
<tr>
<td></td>
<td>Multiple paper</td>
<td>1 metal</td>
<td>1 plastic, 1 stone</td>
<td>Multiple metal, stone, glass</td>
</tr>
</tbody>
</table>
Sample Performance Space Arrangement (duo)

<table>
<thead>
<tr>
<th>1 metal (ex: vibraphone)</th>
<th>1 glass, 1 stone (ex: water glass, large stone)</th>
<th>1 metal, 1 plastic (ex: brake drum, plastic drop cloth)</th>
</tr>
</thead>
<tbody>
<tr>
<td>(empty)</td>
<td>Music stand alone</td>
<td>(empty)</td>
</tr>
<tr>
<td>Multiple paper (ex: several books)</td>
<td>(empty)</td>
<td>Multiple paper, glass, plastic (ex: ream of paper, glass chimes, plastic “granite” blocks, etc.)</td>
</tr>
</tbody>
</table>

Score Notation/Basic Performance Method

The score contains a variety of graphic images that are interpreted by the performer(s); these interpretations should not only account for the types of graphics used, but also how they relate to each other (their arrangement on the page). The graphics are not specific with regard to performance output. Over the course of the piece, players should attempt to express each the “essence” of each graphic image through varied interpretations of each, though repetition may be utilized as part of specific interpretations. Performers should not limit themselves to traditional performance practice; graphics might suggest sounds, visual gestures, or other actions. In sectors that contain instruments, the performer(s) should attempt to incorporate all instruments in their interpretation, though if a sector contains only a music stand, the performer(s) should use their body, voice, music stand, and/or any performance space features that are present in that sector. **Player(s) are encouraged to be creative and explore!** Some pages contain transformation indicators that appear as three different gray letters: I, T, and/or O (standing for “imitate,” “transform,” and “oppose” respectively). These indicate how one should react to external stimuli such as present or past actions and/or sounds presented by the other player, yourself, and/or the electronics. The placement of these letters with respect to the score graphics should become part of one’s interpretation. Each score page has a numeric and alphabetic label in the upper left corner that will be addressed shortly.

Timing

The timing of each interpretation is determined by individual performers. This typically, though not always, spans the amount of time necessary to articulate the essential character of each page. In some cases, it may be desirable to focus one’s
interpretation on a single aspect of the score, and gradually integrate other aspects, with the interpretation of the total page as a goal; other times one may choose to interpret the whole page at once; other temporal possibilities for interpretations exist (shifting focus for example). Once a page interpretation has been completed, that performer must travel to an adjacent sector containing a music stand. Upon reaching the new music stand, the performer must choose a score page that shares either a number or letter label with the previously performed page. For example, if one has just finished a page labeled “3A,” the next page to be performed should have a “3” or an “A” (or both) as part of its label. Two performers may not occupy the same sector simultaneously. The total length of the performance is left to the discretion of the performers.

**Duo versus Trio Versions**

The duo and trio versions use the same score packets (groups of three pages), and performance instructions outlined above. As discussed in the “Performance Space Segmentation” section, the two versions differ in the type of grid used, and therefore the possible number of full/empty sectors. Additionally, the duo/trio performances will offer different stimuli for the performer(s) to respond to when addressing the transformation indicators (I, T, O) in the score. In the duo version, the performer must address either his/her previous actions/sounds and/or the sounds presented by the electronics; in the trio, the performers also can respond actions/sounds created by each other.
Electronics

The electronics performer will use MAX/MSP, a software package with a graphical interface that facilitates real-time synthesis and processing. Nine MAX/MSP patches (discrete panels designed by the user/composer to enable specific synthesis and/or processing tasks) and a copy of MAX/MSP Runtime 4.5 (a freeware program that allows one to perform with but not edit MAX/MSP patches) are included on a CD-ROM that is provided with this piece’s performance materials. Each patch is designed to allow the performer to manipulate one or more predetermined sound files (see below) in various ways during a performance. The electronics performer should use one patch at a time (a tenth patch is provided that facilitates recording; this patch may be used throughout entire rehearsals and performances while another patch is in use).

Each patch should be treated in the same way as the percussionist(s) treat each sector of the performance space. Individual patches are associated with a packet of three score pages, and the manipulation of a patch (described below) should result from the electronics operator’s interpretation of a single score page. For example, when faced with a patch featuring a single band-pass filter and a score page containing a large oscillating line and a small stream of numbers, the electronics operator might manually oscillate the total volume output of the signal while randomizing the central frequency of the filter. The movement from one patch to the next should imitate the percussionist(s) movement between sectors in timing and choice of score pages. The electronics performer determines the length of use of a particular patch based on his/her interpretation of the given score page; upon accessing a new patch, the associated score page should be chosen to match either the number or letter designation of the previously used page. The electronics operator does not use score pages with transformation indicators (I, T, O), though the percussionist(s) can (and should) respond to the sounds produced by the electronics when addressing this facet of their score.

Rehearsals and performances should be recorded whenever possible (using the tenth patch that facilitates recording, or any other means); these recordings will serve as the source material for live processing by the electronics operator. The rehearsal/performance recordings may be shortened or volume-adjusted in any editing program prior to their usage in MAX/MSP.

Each patch requires a significant amount of customization (resulting from the operator’s interpretation of the score) prior to the execution (sound playback) of that patch, as well as potentially during playback. For example, one might configure a random number generator, breakpoint file, and/or use a graphic slider to tailor the processes applied to the sound file(s) including filtering, delay, and ring modulation. The process of configuring each patch will take a small amount of time; this punctuation between electronics entrances is desirable and encouraged. All patches are explained below, and ample labels are present within each patch explaining what modes of interaction are available.
Basic Interfaces

There are a few interfaces that are used throughout this and many other pieces that utilize MAX/MSP. These will be addressed prior to explaining individual patches.

Toggle – This box activates/deactivates an object that it is connected to (it might for example start a random number generator or play a sound file). Clicking a toggle once turns it on and makes an X appear in the box; clicking a second time turns it off and removes the X.

Button – A button functions similarly to a toggle, except that it simply initiates a process when clicked, and cannot be turned off. In this work, buttons are used only to begin finite processes such as breakpoint functions.

Number box – Number boxes are used in two ways in this piece. Primarily, they are used to provide the electronics performer with the numerical output of various objects or processes such as random number generators, sliders, breakpoint functions, and so forth. However, one can also type a numerical value into a number box (if the box contains a decimal point, it is considered a “float number box” and will accept non-integers; if the the box does not contain a decimal point it is an “int number box” and will only accept integers). Hitting the enter key after entering a value in a number box will send that value to any object to which it is connected.

Message boxes – Message boxes (boxes which can contain text or numbers) are used throughout this piece’s MAX/MSP patches. Clicking on a message box sends its data to any object to which it is connected. The message boxes pictured to the left are frequently used in this piece’s patches to select sound files and output channels (in these cases, ample labels are provided to indicate what each message box selects). The boxes containing “clear,” “open,” “pause,” “resume,” and other messages will be discussed shortly.

Slider – Sliders are manually controlled objects that output a range of numbers; they can be manipulated prior to the initiation of sound, or in real-time during playback. Clicking on the slider and dragging will smoothly transition between numerical values while clicking anywhere on the object with the mouse will immediately move the slider to that position.
Breakpoint function editor – Breakpoint function editors allow one to design the trajectory of a specific process over a given span of time (x axis = time; y axis = numerical values at each point in time). Clicking within these boxes will add points that are connected linearly; points can be moved by clicking on them and dragging. The “clear” message box that appears above each breakpoint function editor will clear all points. The range of each breakpoint function is predetermined depending on what process/facet is being designed. The domain (total time) is user determined. In this piece, all breakpoint functions within a single patch are temporally coordinated, so a single number box exists in each patch where one can enter (type) the domain for all breakpoints in seconds to three decimal points. Note that typing the domain and hitting the enter key will initiate all breakpoint functions; if this is not desired, simply type the domain value without hitting the enter key. Changing the breakpoint domain will not affect the shapes present in the breakpoint function editors, they will simply be output at different speeds. Adjacent to the number box that determines the breakpoint function domains is a button which will initiate all breakpoint functions (similar to hitting the enter key after entering the breakpoint domain); this causes the breakpoint functions to smoothly output all of the Y-axis values over the specified time.

Accessing Sound Files
All nine patches allow sound file processing and access these files in the same way. Learning how to open and use these files in one patch will allow you to do the same in all others.

- The “open” message box allows you to select the file that you wish to manipulate. Clicking on it facilitates navigation through the computer, similar to most other applications.
- To initiate playback from the beginning of the file, click the toggle on at the far left. Clicking it again will stop the file’s playback.
- The “pause” and “resume” message boxes are fairly intuitive; the former pauses playback and the latter resumes playback from the point where the file was paused.
- The number box located at the far right allows access to any point within a sound file. Simply click on the box, type the point in seconds (up to three decimal places) where you want the sound file to begin playback, and hit enter; the file will begin playback from the specified point.
- To change the selected sound file at any point, click open and locate a new file. To initiate its playback, it is necessary to deactivate and reactivate the toggle (click to remove the X, and then click again to reactivate).
Methods of Data Manipulation

This piece uses six methods for manipulating data that directly affect sonic output. Each will be discussed at length here and referenced briefly throughout the remaining instructions. Note that number boxes are adjacent to all data manipulations to provide the performer with output information.

- **Manual sliders** – Sliders are used frequently to directly manipulate data (such as volume, delay time, and so forth). These function exactly as described previously.

- **Breakpoint functions** – Breakpoint functions work as described above, and allow the user to determine the trajectory of a single piece of data (such as volume, delay time, and so forth). These functions are of finite length specified in a number box. Y-axis values at the beginning and end of each breakpoint function will automatically be inserted upon initiation of the function to ensure smooth playback and manipulation.

- **Random number generators (4)** – Random number generators are utilized in four different formats. A toggle labeled “Initiate random number generation” is present in all files, and globally starts and stops all random number generators. Since random number generation is a perpetual process (whose data can be used selectively through other means), it is advisable to initiate generation while preparing a patch and let the process continue throughout the use of that patch.
  
  - **Random/regular/smooth** – This format generates numbers at a regular interval specified by a slider, and smoothly transitions (glisses) from one value to the next. A number box adjacent to the slider indicates how often numbers will be generated, and a number box below the slider displays the random number generator’s output.
  
  - **Random/regular/abrupt** – This format generates numbers at a regular interval specified by a slider, and discretely transitions from one value to the next (this sometimes results in subtle clicking). A number box adjacent to the slider indicates how often numbers will be generated, and a number box below the slider displays the random number generator’s output.
  
  - **Random/random/smooth** – This format generates numbers at random intervals, and smoothly transitions (glisses) from one value to the next. A number box below the slider displays the random number generator’s output.
  
  - **Random/random/abrupt** – This format generates numbers at random intervals, and discretely transitions from one value to the next (this sometimes results in subtle clicking). A number box below the slider displays the random number generator’s output.
**Recording**

The file “Record.pat” can be used to record rehearsals and performances (any alternate method for accomplishing this task is also acceptable). This patch contains three interactive aspects, two toggles and an “open” message box.

- First click the uppermost toggle, initiating the program’s analog to digital converter which imports sound from the computer’s soundcard (some customized configuration may be necessary to route microphone input through the computer’s soundcard).
- Next, click the “open” message box. This will allow you to select a filename, destination, and file format (wav and aif are most common, the former for PC and the latter for Mac platforms) for the recorded sounds.
- Finally, click the lower toggle to initiate recording. When finished, click this toggle again to stop recording; the recording will be located and named as specified via the “open” message box.
**Ring Modulation**

The file “Ring.pat” features ring modulation processing. The user can modulate one sound file against another, or against an oscillator whose frequency can be manipulated.

- Begin by opening Sound files 1-4 using the interface located in the lower left section of the screen.
- Using the message boxes containing numbers, select the initial original and modulating signals (these can be changed during performance). Selecting the “None” message box for one of these signals will result in the unaltered playback of the other signal. Selecting “None” for both signals will result in no sound output.
- If using the oscillator as the modulating signal, the right side of the screen provides methods for manipulating its frequency. Click the numbered message box that corresponds to the initial manipulation you wish to use (the manipulation processes are described previously in the “Methods of Data Manipulation” section). Make any necessary preparations to the sliders and breakpoint function editor, and initiate random number generation.
- Adjust the original signal volume, modulating signal volume, and total volume sliders; it is important to note that volume output is greater for non-modulated and oscillator-modulated sound files than when two sound files are modulated.
- Use the toggle in the upper left corner of the screen to turn audio output on.
- Use the sound file toggles to begin sound file playback (one can either start individual files or all).
- Once all of this has been accomplished, one can switch between signals, open and start new sound files, adjust how the oscillator frequency is manipulated, and adjust the volume of each signal and sound output.
Choose how to manipulate oscillator frequency

- Breakpoint
- Manual slider
- Random/regular/smooth
- Random/regular/abrupt
- Random/random/smooth
- Random/random/abrupt

Initiate random number generation

Determine total length for breakpoint file

Initiate breakpoint execution

<table>
<thead>
<tr>
<th>Start all soundfiles</th>
<th>Turn audio on and off (x = on)</th>
<th>Choose the original signal</th>
<th>Choose the modulating signal</th>
</tr>
</thead>
<tbody>
<tr>
<td>Soundfile 1</td>
<td></td>
<td>Soundfile 1</td>
<td>Soundfile 3</td>
</tr>
<tr>
<td>Soundfile 2</td>
<td></td>
<td>Soundfile 2</td>
<td>Soundfile 4</td>
</tr>
<tr>
<td>Soundfile 3</td>
<td></td>
<td>None</td>
<td>Oscillator</td>
</tr>
<tr>
<td>Soundfile 4</td>
<td></td>
<td></td>
<td>None</td>
</tr>
</tbody>
</table>

Choose the original signal:
1. Soundfile 1
2. Soundfile 2
3. None

Choose the modulating signal:
1. Soundfile 3
2. Soundfile 4
3. Oscillator
4. None

Total volume
Orig. sig. vol.
Mod. sig. vol.

Start all soundfiles

Soundfile 1
- open
- pause
- resume
- 0.

Soundfile 2
- open
- pause
- resume
- 0.

Soundfile 3
- open
- pause
- resume
- 0.

Soundfile 4
- open
- pause
- resume
- 0.

Ring.pat
Single Filter

The file “Single Filter.pat” features processing by a single band-pass filter. The user can manipulate volume, center frequency, and bandwidth (Q-factor).

- Begin by opening Sound files 1 and 2 using the interface located in the upper left section of the screen. Use the message boxes below “Sound file 1” to select which file to filter initially.
- Use the message boxes containing the numbers 1 through 6 in the middle of the screen to initially determine how to manipulate the filter’s volume, center frequency, and bandwidth (the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Make any necessary preparations to the sliders and breakpoint function editor, and initiate random number generation.
- Use the toggle in the upper middle of the screen to turn audio output on.
- Use the sound file toggles to begin playback.
- Once all of this has been accomplished, one can switch between sound files, open and start new sound files, and adjust how the volume, center frequency, and bandwidth are manipulated.
Choose the soundfile to be played back and processed:
1. None
2. Soundfile 1
3. Soundfile 2

Choose how to manipulate volume:
1. Breakpoint
2. Manual slider
3. Random/ regular/smooth
4. Random/ regular/abrupt
5. Random/ random/smooth
6. Random/ random/abrupt

Choose how to manipulate center frequency:
1. Breakpoint
2. Manual slider
3. Random/ regular/smooth
4. Random/ regular/abrupt
5. Random/ random/smooth
6. Random/ random/abrupt

Choose how to manipulate bandwidth:
1. Breakpoint
2. Manual slider
3. Random/ regular/smooth
4. Random/ regular/abrupt
5. Random/ random/smooth
6. Random/ random/abrupt

Single filter.pat
Single Delay

The file “Single Delay.pat” features delay and feedback processing. The user can manipulate the volume of the un-delayed and delayed sound files (the former is heard in the left channel, the latter in the right), delay time, and feedback strength.

- Begin by opening Sound files 1 and 2 using the interface located in the upper left section of the screen. Use the message boxes below “Sound file 1” to select which file to delay initially.
- Use the message boxes containing the numbers 1 through 6 in the middle of the screen to initially determine how to manipulate the delayed and un-delayed volume, delay time, and feedback strength (the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Make any necessary preparations to the sliders and breakpoint function editor, and initiate random number generation.
- Use the toggle in the upper middle of the screen to turn audio output on.
- Use the sound file toggles to begin sound file playback.
- Once all of this has been accomplished, one can switch between sound files, open and start new sound files, and adjust how the volume, delay time, and feedback strength are manipulated.
Single Delay.pat
Multiple Filters
The file “Multiple Filters.pat” features processing of one to six sound files by up to six band-pass filters. The user can manipulate volume, center frequency, and the bandwidth (Q-factor) for each filter.

- Begin by opening Sound files 1 through 7 using the interface located on the left section of the screen.
- The six filters are labeled. Use the message boxes containing the numbers 0 through 7 to select which file each filter will process (selecting 0 will negate the use of that filter). Note that it is possible to filter a single sound file six different ways, or to filter six sound files simultaneously.
- Using the message boxes containing the numbers 0, 1, and 2 to select which channel each filter will initially output to (1 = left, 2 = right).
- Make any necessary preparations to the sliders and breakpoint function editors, and initiate random number generation (note that each filter uses the same type of manipulation for all three of its variables; the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Use the toggle in the upper left of the screen to turn audio output on.
- Use the sound file toggles to begin playback.
- Once all of this has been accomplished, one can switch between sound files, filters, and output channels, open and start new sound files, and adjust how the volume, center frequency, and bandwidth are manipulated.
Multiple Filters.pat
Multiple Delays

The file “Multiple Delays.pat” features processing of one to six sound files by up to six delays. The user can manipulate volume, delay time, and feedback strength for each delay.

- Begin by opening Sound files 1 through 7 using the interface located on the left section of the screen.
- The six delays are labeled. Use the message boxes containing the numbers 0 through 7 to select which file each filter will process (selecting 0 will negate the use of that delay). Note that it is possible to delay a single sound file six different ways, or to delay six sound files simultaneously.
- Using the message boxes containing the numbers 0, 1, and 2 to select which channel each delay will initially output to (1 = left, 2 = right).
- Make any necessary preparations to the sliders and breakpoint function editors, and initiate random number generation (note that each delay uses the same type of manipulation for all three of its variables; the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Use the toggle in the upper left of the screen to turn audio output on.
- Use the sound file toggles to begin playback.
- Once all of this has been accomplished, one can switch between sound files, delays, and output channels, open and start new sound files, and adjust how the volume, delay time, and feedback strength are manipulated.
Filter-Delay Combination

The file “Filter-Delay.pat” features processing of one sound file by a band-pass filter and delay. The user can manipulate volume, center frequency, bandwidth, delay time, and feedback strength.

- Begin by opening Sound files 1 and 2 using the interface located in the upper left section of the screen. Use the message boxes below “Sound file 1” to select which file to delay initially.
- Use the message boxes containing the numbers 1 through 6 in the middle and bottom of the screen to initially determine how to manipulate the filtered/un-delayed and filtered/delayed volumes (spatialization is similar to the Single Delay patch), delay time, and feedback strength. Use the message boxes containing the numbers 1 through 6 in the middle of the screen to initially determine how to manipulate the delayed and un-delayed volume, delay time, and feedback strength. (the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Make any necessary preparations to the sliders and breakpoint function editors, and initiate random number generation.
- Use the toggle in the upper middle of the screen to turn audio output on.
- Use the sound file toggles to begin sound file playback.
- Once all of this has been accomplished, one can switch between sound files, open and start new sound files, and adjust how the volume, center frequency, bandwidth, delay time, and feedback strength are manipulated.
Filter-Delay.pat
Ring Modulation-Filter Combination

The file “Ring-Filter.pat” features ring modulation and band-pass filter processing. The user can modulate one sound file against another, or against an oscillator whose frequency can be controlled using several methods while manipulating the filter’s volume, center frequency, and bandwidth (Q-factor).

- Begin by opening Sound files 1-4 using the interface located in the lower left section of the screen.
- Using the message boxes containing numbers, select the initial original and modulating signals (these can be changed during performance). Selecting the “None” message box for one of these signals will result in the unaltered playback of the other signal. Selecting “None” for both signals will result in no sound output.
- If using the oscillator as the modulating signal, click the numbered message box that corresponds to the initial frequency manipulation you wish to use (the manipulation processes are described previously in the “Methods of Data Manipulation” section). Make any necessary preparations to the “Ring Modulation” sliders and breakpoint function editor, and initiate random number generation.
- Adjust original and modulating signal volumes; it is important to note that volume output is greater for non-modulated and oscillator-modulated sound files than when two sound files are modulated.
- In the “Filter” portion of the screen, use the message boxes containing the numbers 0, 1, and 2 to select which channel each filter will initially output to (1 = left, 2 = right).
- Make any necessary preparations to the “Filter” sliders and breakpoint function editors, and, if not previously completed, initiate random number generation (note that each filter uses the same type of manipulation for all three of its variables; the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Use the toggle in the upper middle of the screen to turn audio output on.
- Use the “Determine max output” sliders in the middle of the screen to set initial volume levels for each channel (1 is normal).
- Use the sound file toggles to begin sound file playback (one can either start individual files or all).
- Once all of this has been accomplished, one can switch between signals, open and start new sound files, adjust how the oscillator frequency is manipulated, adjust the volume of each signal and sound output, switch between filters and output channels, and adjust how the filter volume, center frequency, and bandwidth are manipulated.
Ring Modulation-Delay Combination

The file “Ring-Delay.pat” features ring modulation and delay processing. The user can modulate one sound file against another, or against an oscillator whose frequency can be controlled using several methods while manipulating volume, delay time, and feedback strength.

- Begin by opening Sound files 1-4 using the interface located in the lower left section of the screen.
- Using the message boxes containing numbers, select the initial original and modulating signals (these can be changed during performance). Selecting the “None” message box for one of these signals will result in the unaltered playback of the other signal. Selecting “None” for both signals will result in no sound output.
- If using the oscillator as the modulating signal, click the numbered message box that corresponds to the initial frequency manipulation you wish to use (the manipulation processes are described previously in the “Methods of Data Manipulation” section). Make any necessary preparations to the “Ring Modulation” sliders and breakpoint function editor, and initiate random number generation.
- Adjust original and modulating signal volumes; it is important to note that volume output is greater for non-modulated and oscillator-modulated sound files than when two sound files are modulated.
- In the “Delay” portion of the screen, use the message boxes containing the numbers 0, 1, and 2 to select which channel each filter will initially output to (1 = left, 2 = right).
- Make any necessary preparations to the “Delay” sliders and breakpoint function editors, and, if not previously completed, initiate random number generation (note that each delay uses the same type of manipulation for all three of its variables; the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Use the toggle in the upper middle of the screen to turn audio output on.
- Use the “Determine max output” sliders in the middle of the screen to set initial volume levels for each channel (1 is normal).
- Use the sound file toggles to begin sound file playback (one can either start individual files or all).
- Once all of this has been accomplished, one can switch between signals, open and start new sound files, adjust how the oscillator frequency is manipulated, adjust the volume of each signal and sound output, switch between delays and output channels, and adjust how the volume, delay time, and feedback strength are manipulated.
Ring-Delay.pat
Ring Modulation-Filter-Delay Combination

The file “Ring-Delay.pat” features ring modulation, band-pass filter, and delay processing. The user can modulate one sound file against another, or against an oscillator whose frequency can be controlled using several methods while manipulating volume, center frequency, bandwidth (Q-factor), delay time, and feedback strength.

- Begin by opening Sound files 1-4 using the interface located in the lower left section of the screen.
- Using the message boxes containing numbers, select the initial original and modulating signals (these can be changed during performance). Selecting the “None” message box for one of these signals will result in the unaltered playback of the other signal. Selecting “None” for both signals will result in no sound output.
- If using the oscillator as the modulating signal, click the numbered message box that corresponds to the initial frequency manipulation you wish to use (the manipulation processes are described previously in the “Methods of Data Manipulation” section). Make any necessary preparations to the “Ring Modulation” sliders and breakpoint function editor, and initiate random number generation.
- Adjust original and modulating signal volumes; it is important to note that volume output is greater for non-modulated and oscillator-modulated sound files than when two sound files are modulated.
- Use the message boxes containing the numbers 1 through 6 in the middle and bottom of the screen to initially determine how to manipulate the delayed and undelayed volumes, center frequency, bandwidth, delay time, and feedback strength (the manipulation processes are described previously in the “Methods of Data Manipulation” section).
- Make any necessary preparations to the “Filter” and “Delay” sliders and breakpoint function editors, and, if not previously completed, initiate random number generation (note that each delay uses the same type of manipulation for all three of its variables).
- Use the toggle in the upper middle of the screen to turn audio output on.
- Use the sound file toggles to begin sound file playback (one can either start individual files or all).
- Once all of this has been accomplished, one can switch between signals, open and start new sound files, adjust how the oscillator frequency is manipulated, adjust the volume of each signal, and adjust how the volume, center frequency, bandwidth, delay time, and feedback strength are manipulated.
Performance Score
Single Filter.pat background

Choose the soundfile to be played back and processed
1. None
2. Soundfile 1
3. Soundfile 2

Choose how to manipulate volume
1. Breakpoint
2. Manual slider
3. Random/regular/abrupt
4. Random/regular/smooth
5. Random/random/abrupt
6. Random/random/smooth

Choose how to manipulate center frequency
1. Breakpoint
2. Manual slider
3. Random/regular/abrupt
4. Random/regular/smooth
5. Random/random/abrupt
6. Random/random/smooth

Choose how to manipulate bandwidth
1. Breakpoint
2. Manual slider
3. Random/regular/abrupt
4. Random/regular/smooth
5. Random/random/abrupt
6. Random/random/smooth

Soundfile 1
- open
- pause
- resume
- seek $1
- selector~ 2

Soundfile 2
- open
- pause
- resume
- seek $1
- selector~ 2

Initiate random number generation
- send bang

Initiate breakpoint execution(s)
- p bkpset
- p bkpupvol

Determine total length for all breakpoint files
- send bang
- 2 - sldr

Turn audio output on and off (x = on)
- dac~
- p ranransm
- 0.

- p ranranab
- 0.

- p ranransm2
- 0.

- p ranranab2
- 0.

- p ranransm3
- 0.

- p ranranab3
- 0.
Single Filter.pat subpatches

bkset
- receive bangrand
- setdomain $1
- send bkptdomain
- send final
- line
- send bkptstart

b kp upvol
- receive bkptstart
- receive bkptdomain
- receive final
- send bkptdomain
- setdomain $1
- line
- send bkptstart

ranregab
- receive bangrand
- metro 1000
- random 50
- 0.1

ranregsm
- receive bangrand
- metro 1000
- random 50
- 0.1

ranransm
- receive bangrand
- metro 1000
- random 50
- 0.1

ranranab
- receive bangrand
- metro 1000
- random 50
- 0.1
Single Filter.pat subpatches continued

ranregsm2

ranregab2

ranransm2

ranranab2

ranregsm3

ranregab3
Single Filter.pat subpatches continued

receive bangrand

metro 50
random 5000

metro 1000

random 100 0

line

---

receive bangrand

metro 50
random 5000

metro 1000

random 100
Single Delay.pat background

Choose the soundfile to be played back and processed

Choose how to manipulate the volume of the unprocessed soundfile

Initiate random number generation

Determine total length for all breakpoint files

Initiate breakpoint execution(s)

Turn audio output on and off (x = on)

Choose how to manipulate the delay time

Choose how to manipulate the feedback strength
Single Delay.pat subpatches

bkupvol

ranregsm

ranregab

ranransm

ranranab

bkpset
Single Delay.pat subpatches continued

ranregsm2

ranregab2

ranransm2

ranranab2

ranregsm3

ranregab3
Multiple Filters.pat subpatches

sel1

sel2

ap1

ap2
Multiple Filters.pat subpatches continued

**ranregsm1**
- receive bangrand
- metro 1000
- random 50
- line
- 0.1
- send v2

**ranregsm2**
- receive bangrand
- metro 1000
- random 7970
- line
- +30
- +30
- send q2

**ranregsm3**
- receive bangrand
- metro 1000
- random 100
- line
- send q2

**sel3**
- receive~ sf1
- receive~ sf2
- receive~ sf3
- receive~ sf4
- receive~ sf5
- receive~ sf6
- receive~ sf7
- selector~ 7

**ap3**
- receive v3
- receive cf3
- receive q3
- gate~ 2
- reson~
Multiple Filters.pat subpatches continued

**ranregab1**

- `receive bangrand`
- `metro 1000`
- `random 50`
- `-0.1`
- `send v3`

**ranregab2**

- `receive bangrand`
- `metro 1000`
- `random 7970`
- `+30`
- `send cf3`

**ranregab3**

- `receive bangrand`
- `metro 1000`
- `random 100`
- `send q3`

**sel4**

- `selector~ 7`
- `receive~ sf1`
- `receive~ sf2`
- `receive~ sf3`
- `receive~ sf4`
- `receive~ sf5`
- `receive~ sf6`
- `receive~ sf7`
Multiple Filters.pat subpatches continued

receive bangrand

metro 500
random 5000

metro 1000
random 50
line

metro 1000
random 7970

metro 500
random 5000

metro 1000
random 100
line

send v4
send cf4
send q4

+30

reson~ 1. 1000 1

ap4

gate~ 2
Multiple Filters.pat subpatches continued

```
selector~ 7
receive~ sf1
receive~ sf2
receive~ sf3
receive~ sf4
receive~ sf5
receive~ sf6
receive~ sf7
```

sel5
Multiple Filters.pat subpatches continued

receive bangrand

metro 500
random 5000
metro 1000
random 50

gate~ 2

send v5
send cf5
send q5

reson~ 1.1000 1

0.1
+30
0.
0.
0.

random 50
random 1000
random 100

send q5

ap5
Multiple Filters.pat subpatches continued

selector~ 7
receive~ sf1  receive~ sf2  receive~ sf3  receive~ sf4  receive~ sf5  receive~ sf6  receive~ sf7

selector~ 7

receive bkptime

seldomain $1
line

receive bang

0.0.
$1.0.

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1

 receive bkptime

  1000

 receive v6
 receive cf6
 receive q6
 reson~ 1.1000 1
Multiple Delays.pat subpatches

**sel1**

**sel2**

**ap1**

**ap2**
Multiple Delays.pat subpatches continued

```
milan 1000
random 50
line 0
* 0.1
receive bangrand
send v2

milano 1000
random 99
line 0
* 0.01
receive bangrand
send f2

milan 1000
random 10000
line 0
receive bangrand
send d2
```

```
ranregsm1     ranregsm2     ranregsm3
selector~ 7
receive~ sf1
receive~ sf2
receive~ sf3
receive~ sf4
receive~ sf5
receive~ sf6
receive~ sf7
```

```
receiver~ sf1
receiver~ sf2
receiver~ sf3
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receiver~ sf1
receiver~ sf2
receiver~ sf3
receiver~ sf4
receiver~ sf5
receiver~ sf6
receiver~ sf7
```

```
receiver~ sf1
receiver~ sf2
receiver~ sf3
receiver~ sf4
receiver~ sf5
receiver~ sf6
receiver~ sf7
```
Multiple Delays.pat subpatches continued

```
allpass~ 10000 1000. 0.
~ 1.
gate~ 2
receive v3
receive f3
receive d3
*
metro 1000
random 50
*
0.1

receive bangrand
send v3
```

```
metro 1000
random 10000

receive bangrand
send d3
```
Multiple Delays.pat subpatches continued

```
multiple delays

receive bangrand
metro 1000
random 99
-0.01
send f3

receive~ sf1
receive~ sf2
receive~ sf3
receive~ sf4
receive~ sf5
receive~ sf6
receive~ sf7
selector~ 7
ranregab3
sel4
```
Multiple Delays.pat subpatches continued

```
receive bangrand

metro 500
random 5000

metro 1000
random 50

metro 1000
random 5000

metro 1000
random 10000

0.1

line

allpass~ 10000 1000. 0.

send v4
send d4
send f4

ap4
```
sel5
Multiple Delays.pat subpatches continued

sel6

ap6
Filter-Delay.pat subpatches continued

ranregsm2

ranregab2

ranransm2

ranranab2

ranregsm3

ranregab3
Filter-Delay.pat subpatches continued

- ranransm3
  - receive bangrand
  - metro 50
  - random 5000
  - metro 1000
  - random 100
  - 0
  - line
- ranranab3
  - receive bangrand
  - metro 50
  - random 5000
  - metro 1000
  - random 100
- ranregsm4
  - receive bangrand
  - 10
  - metro 1000
  - random 50
  - 0
  - line
  - 0.1
- ranregab4
  - receive bangrand
  - 10
  - metro 1000
  - random 50
  - 0.1
  - line
  - 0.1
- ranransm4
  - receive bangrand
  - metro 50
  - random 5000
  - metro 1000
  - random 50
  - 0
  - line
  - 0.1
- ranranab4
Filter-Delay.pat subpatches continued

ranregsm5

ranregab5

ranransm5

ranranab5

ranregsm6

ranregab6
Filter-Delay.pat subpatches continued

receive bangrand
  
  metro 50
  random 5000

metro 1000
  random 99
  0
  line
  0.01

ranransm6

receive bangrand
  
  metro 50
  random 5000

metro 1000
  random 99
  0.01

ranranab6
Ring-Filter.pat subpatches

bkpt

receive bkptime
-1000

setdomain $1

receive bang

0, $1 $1

line

0, 0, $1 0.

ranregsm

receive bangrand
+30
random 7970
metro 1000

line

ranregab

receive bangrand
+30
random 7970
metro 1000

Ring-Filter.pat subpatches continued

```
receive bangrand
  metro 1000
  random 100
  line
  0
  receive bangrand
  send q2
  ranregsm3

receive v3
  receive cf3
  receive q3
  reso~
  gate~ 2
  ap3

receive bangrand
  metro 1000
  random 50
  * 0.1
  send v3
  ranregab1
```
Ring-Filter.pat subpatches continued

```
receive bangrand
metro 1000
random 7970
+30
send cf3
```

```
receive bangrand
metro 1000
random 100
send q3
```

ranregab2  ranregab3
Ring-Filter.pat subpatches continued

receive v6
receive q6
receive q6
receive q6
reson~ 1: 1000 1

ap6
Ring-Delay.pat subpatches continued

**ranransm**

**ranranab**

**ap1**
Ring-Delay.pat subpatches continued

allpass~ 10000 1000. 0.

~ 1.

receive d2
receive v2

~ 1.

gate~ 2

random 50

send v2

random 10000

send d2
Ring-Delay.pat subpatches continued

- receive bangrand
- metro 1000
- random 99
- * 0.01
- line
- send f2

- receive d3
- allpass~ 10000 1000. 0.
- receive f3
- *~ 1.
- allpass~ 10000 1000. 0.
- receive v3
- gate~ 2

- receive bangrand
- metro 1000
- random 50
- * 0.1
- send v3
Ring-Delay.pat subpatches continued

ranregab2

ranregab3
Ring-Delay.pat subpatches continued

receive bangrand

metro 500
random 5000

random 50

line

metro 1000
random 50

metro 500
random 5000

metro 1000
random 10000
random 99

line

0.1

send v4
send d4
send f4

allpass~ 10000 1000. 0.

c-1

gate~ 2

ap4
Ring-Delay.pat subpatches continued

receive bangrand

metro 1000
random 50
random 5000
metro 500
random 5000
metro 500
random 5000
metro 500
random 5000
metro 500
random 5000
metro 500
random 5000

~ 0.1

send v5
send d5
send f5

allpass~ 10000 1000. 0.

~ 1

gate~ 2

ap5
Ring-Delay.pat subpatches continued

receive v6

* 1000.

receive d6

receive f6

allpass~ 10000 1000. 0.

~ 1.

gate~ 2

ap6
Ring-Filter-Delay.pat subpatches

bkipset

receive bangrand
random 7970
metro 1000
random 5000
metro 50

ranregab

bkipvol

receive bangrand
random 7970
10
metro 1000

ranransm

receive bangrand
random 7970
line
+ 30

ranranab
Ring-Filter-Delay.pat subpatches continued

ranregsm1

ranregab1

ranransm1

ranranab1

ranregsm2
Ring-Filter-Delay.pat subpatches continued

- ranregab2
  - receive bangrand
  - random 7970
  - metro 1000
  - random 5000
  - line 0.1
  - 0
  - receive bangrand

- ranransm2
  - receive bangrand
  - random 5000
  - line 0
  - 10
  - receive bangrand

- ranranab2
  - receive bangrand
  - random 5000
  - line 0
  - 10
  - receive bangrand

- ranregsm3
  - receive bangrand
  - random 5000
  - line

- ranregab3
  - receive bangrand
  - random 5000
  - line

- ranransm3
  - receive bangrand
  - random 5000
  - line
Ring-Filter-Delay.pat subpatches continued

ranranab3

ranregsm4

ranregab4

ranransm4

ranranab4

ranregsm5
Ring-Filter-Delay.pat subpatches continued

ranregab5

ranransm5

ranranab5

ranregsm6

ranregab6
Ring-Filter-Delay.pat subpatches continued

receive bangrand
  \|-- metro 50
     \|-- random 5000
     \|-- metro 1000
  \|-- random 99
     \|-- 0
     \|-- line
        \|-- * 0.01

ranransm6

receive bangrand
  \|-- metro 50
     \|-- random 5000
     \|-- metro 1000
  \|-- random 99
     \|-- * 0.01

ranranab6