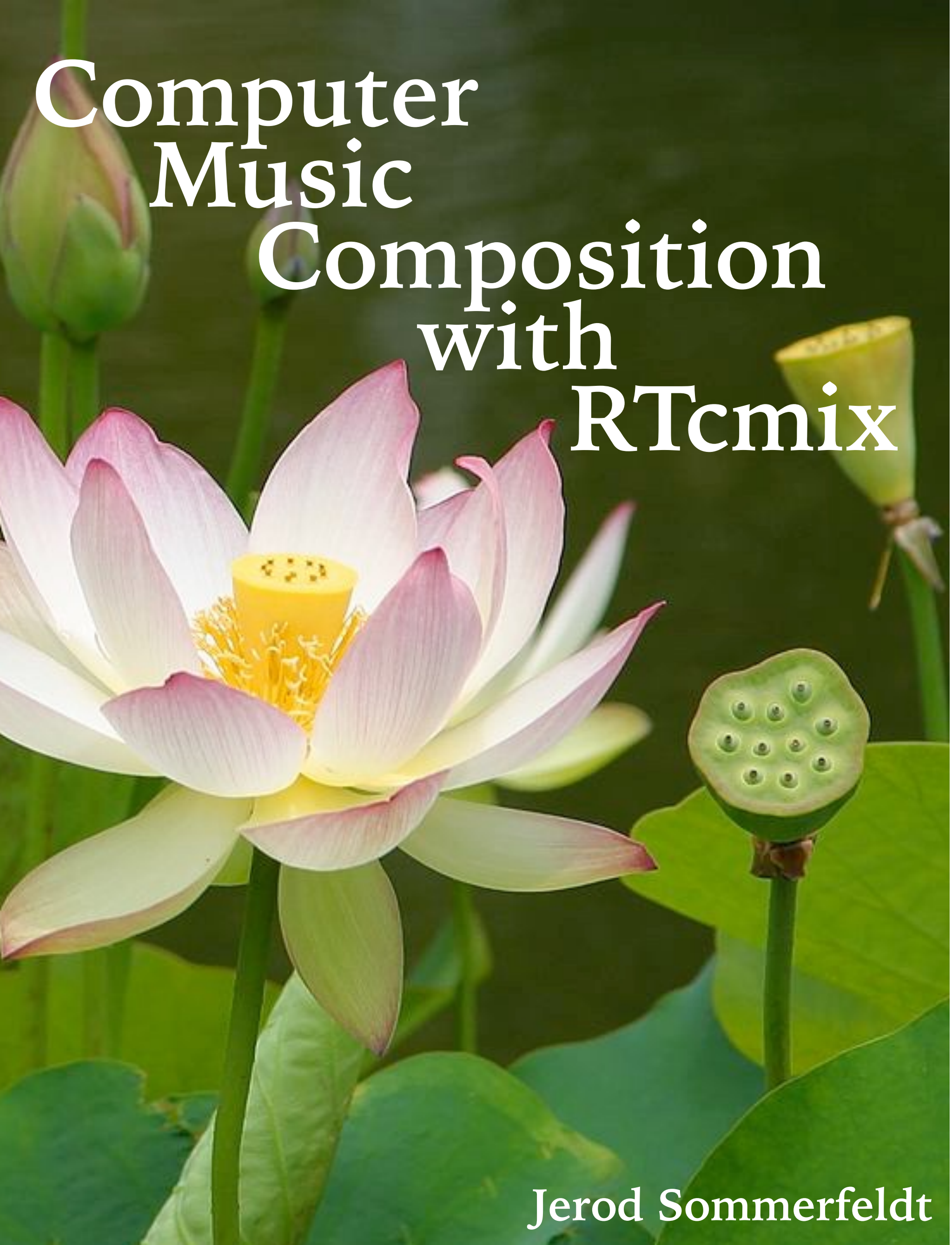
Computer Music Composition with RTcmix



Jerod Sommerfeldt

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This book is dedicated to all past, present, and especially future users of RTcmix.

||Foreword||

In today's fast-paced world of digital technology, composers interested in creating digital music find themselves very far from a shortage of tools to assist them in the realization of their art. As a student, I recall very clearly during a lesson with Mara Helmuth, my teacher at CCM, that I was fast becoming frustrated with the sheer volume of software suites — both free and proprietary — at my disposal. Prior to my arrival in Cincinnati for my DMA, I had already been introduced to electroacoustic and computer music while privately studying composition with Christopher Burns at the University of Wisconsin-Milwaukee, where I had taken a number of classes using the graphical programming software Pure Data, a seminar on algorithmic composition using Scheme and Common Music LISP, and spent two years making music on my laptop with the Milwaukee Laptop Orchestra. With regard to digital audio workstations, I'd taken classes on Digital Performer, owned my own copy of Pro Tools, was working in a studio that used Logic, and was at the time enrolled in a class that used Audacity! With so many options, I was becoming confused about how I could better streamline my work, stop worrying about the idiosyncrasies of each individual tool, and begin focusing solely on the music at hand.

Eventually, I found that I wanted to write music that would allow me to express myself in a sound world limited only by my imagination, would afford me the chance to explore and study the basic tenets of digital signal processing, and would provide me the opportunity to feel as if I was working with my computer

at a deeper level than big-box, proprietary software — with its graphical user interfaces and point-and-click methodology — would allow.

Enter RTcmix.

Here was a program that took the text I typed into my computer and processed the sounds by way of a bash shell! (More on that later.) All of a sudden, I was using my Mac OS X Terminal, learning some cool UNIX commands, constructing loops, conditionals, and all of the things that I yearned for as a budding computer musician who admittedly had zero computer science background. I was finally "under the hood" and tinkering with the engine, freely manipulating exciting sounds using syntax and understanding not only how I made and arranged them, but how I could further sculpt them to my tastes.

Moreover, I was using a program that shares a lineage to the very genesis of computer music and has been utilized by some of the pioneers in the field. RTcmix's authors continue to maintain, update, and expand the program so that it can now be integrated into iPhone and Android applications or Pure Data and Max/MSP. With a supportive community of like-minded musicians and developers, questions about using RTcmix are never more than an email away. In short, RTcmix is a dynamic, exciting tool for anyone interested in creating digital music, be it interactive, electroacoustic, installation work, fixed media, or music that is playable on mobile platforms.

So, why this book?

Outside of the terrific tutorials and documentation on its website — as well as course notes from its authors — RTcmix lacks a concise, all-in-one introductory text that is especially useful for those being introduced to the program for the first time. The pages that follow fill these needs, serving as a comprehensive introduction to the world of RTcmix. Lessons are approached methodologically and take the reader through the basics of installation, explain some very useful

commands in UNIX, demonstrate how to write score files and play them back in its native MINC language and Python, will carefully explain various RTcmix instruments and commands, and will finally look into more advanced topics, such as running RTcmix in a variety of platforms, which will be fun for beginners and useful for all who seek to make exciting computer music.

The mantra of this book is musical creation and is divided into a series of sonatas and interludes. Following a brief prelude that surveys the history of RTcmix, each sonata focuses on a specific task in RTcmix that culminates in a set of suggested directions for composing the reader's own, brief work that together could comprise a new finished piece or suite of pieces. Interludes will cover topics introduced in each sonata and seek to explain those concepts in greater detail.

Nothing is assumed w/r/t prior experience in the world of computer science or computer music. For most of my teaching life, I've had the absolute pleasure of working with beginners who find great joy in learning about computer music using RTcmix from the ground up. While topics such as synthesis, MIDI, Pure Data and Max, and envelopes are discussed, those discussions are not at all comprehensive and may hope is that with the wealth of access to information in our digital world, you'll find it rewarding to pause and further research any topic in this book on your own. Those who are super familiar with Python, for example, will likely find the Python discussions to be rote, but you are of course welcome to further sculpt these RTcmix scripts and examples using your expanded knowledge of the language.

This book and its accompanying score files were all written using an Apple MacBook Pro running Mac OS X 10.10.5 Yosemite. Thus, I will refer the user to the Terminal utility and some useful, ancillary downloads and applications that have been tested and used extensively on the Mac platform. Moreover, I prefer writing my scripts in the Terminal using nano or pico, although alternate approaches to writing your score files will be explored.

And now, an apologia. I am not specifically a computer programmer by training. While I enjoy writing code and have taken extensive classes using computer programming languages for the purposes of making music and am still endlessly fascinated by the sound world I can achieve using lines of text, I know very little about the world of computer science outside of the tiny space I've cultivated for myself. I do know that I am first and foremost and will always be a composer. I just happen to be a composer whose sonic palette and compositional world is enriched by C, Python, RTcmix, Pure Data, Max, Processing, and other digital tools.

Furthermore, this book is not intended to be used as a complete manual for the vast world of digital synthesis and computer music in general. My hope is that the words that follow are written in an inviting, welcoming way for everyone, regardless of prior knowledge or background. I've always appreciated working though texts where the tone is conversational and it feels as if the author is right there next to you, working out the code along with you as you are writing it yourself, so I hope that you find this book a welcoming introduction to this world.

Where details in signal processing procedures are glossed over, my hope is that this text — focusing solely on just a few of the many ways to compositionally approach RTcmix — instills a reliance on self-research for the reader. I often refer to RTcmix's online documentation for further reading and often find it most useful to work with that documentation in an open browser window for quick reference. Moreover, there are far too many really, really great books written by far more capable minds than mine to enlighten and enrich your experience with computer music: Miller Puckette, Curtis Roads, and Charles Dodge come immediately to mind, and a listing of their books and those by others are included at the end of this text.

We'll find that RTcmix can be used in any of a variety of ways. It is still used by default using its own inherent language, MINC, which shares some simi-

larities to C. However, I find that the budding popularity of Python, combined with Python's own expansive set of libraries and functions, greatly enhances the young computer musician's experience with RTcmix. Thus, this book covers both MINC and Python, investigating places where it is necessary to unlock the difference between the two languages and discusses moments where you'll absolutely need to use MINC for your scripts. Suffice it to say that due to its widespread use, learning Python in conjunction with RTcmix will open doors to programming projects far beyond the scope of this program, due in large part to Python's high standing in the world of computer programming.

If you've read this far, you probably know that some thanks are coming and until I actually took the time to sit down and write a book, I never realized just how important it is to give thanks to those who are important in your development. This book would not exist were it not for the invaluable work of two of my teachers, who I am proud to call mentors and friends: Mara Helmuth and Christopher Burns. Brad Garton and John Gibson are two of the authors of RTcmix and their documentation, music, correspondence, and support and encouragement are central to not only me, but to the RTcmix community at-large. Joel Matthys and David McDonnell are two dear friends who share the love I have for this program and our continued correspondence and sharing cool score files and answering each others programming questions makes me endlessly happy and completely and totally grateful. My wife Christi is amazingly supportive of the things that I do. Our two kitties Emma and Oscar would often try desperately to add in their two cents by walking over my keyboard, though they still leave the room when I start composing or making noise.

Perhaps one day our son Foster will work though this book and make his own noisy RTcmix sounds himself.

|| Prelude: A Brief History of RTcmix ||

It is impossible to discuss the history of any generative music language for the computer without first introducing the work of Max Mathews. An engineer, programmer, and musician, Mathews' pioneering work in the digital synthesis of sound began at Bell Laboratories in the 1950s. Interested in the musical possibilities of emerging digital technology, he developed MUSIC I in 1957, a generative program that was able to realize sound using a triangle wave at specified durations, amplitudes, and frequencies. Further refining his series of programs (dubbed MUSIC-N), Mathews completed MUSIC V in 1968, which was an efficient, universal release that met the demands of the fast-evolving technology of computers. With each subsequent release of MUSIC-N, by either Mathews or others in the field who were adapting it for their own use, the notion of portability was being realized. While at first, synthesized sound and computer music was only capable of being executed at Bell Labs, it was soon being disseminated to Princeton, IRCAM, MIT, and Columbia, and today finds itself in the homes of anyone with interest and a computer or tablet device.

Throughout the 1980s, Paul Lansky was at Princeton, working on refining a program he called MIX, which was written using Fortran. Music composed with MIX was realized using punch cards and magnetic tape, and MIX was capable of mixing together various sound files. Lansky decided to add libraries of C functions to the program and make it capable of running on UNIX machines, eventually dubbing his work CMIX. With the versatility of C, users were able to create

dynamic scripts that contained loops and conditional tests to further sculpt and refine their sounds. Moreover, CMIX contained powerful instruments written by Lansky, including linear predictive coding.

CMIX was being further developed as the decade moved along and was ported to a variety of personal computers, including NeXT systems. As computers became more and more portable and powerful, it was becoming of interest to utilize CMIX and other languages in the moment. In 1995, NeXT computers failed, and Brad Garton and Dave Topper — two major contributors to the development of CMIX — began work on a version of the program for the Silicon Graphics platform and eventually Linux. Their work uncovered something quite cool: Following their successful porting to the new platforms, many CMIX instruments were capable of being processed in real-time. They were able to find a solution that directly connected the synthesis engine of CMIX to the powerful digital-to-analog converters that were intrinsic to the new machines of the time. Thus, it was possible to process live sound and manipulate data on-the-fly.

RTcmix was born.

As the 1990s and 2000s moved along to the present, many new contributors came on board for the RTcmix project. Thanks to the efforts of John Gibson and Doug Scott, instruments could be chained together, several new instruments were added to the RTcmix library, scores could be written in Perl or Python, and a body of documentation became available for everyone to peruse online. Mara Helmuth added not only powerful probability functions, but also a stochastic granular synthesis instrument. Running on the command-line, RTcmix was distributed to and maintained by a community of users, who continually added to and tweaked the program for their use. As graphical programming environments began to gain popularity among the computer music community, Luke DuBois and Brad Garton developed an [rtcmix~] object for Max/MSP and Joel Matthys wrote the object for Pure Data, further widening RTcmix's audience. In fact,

RTcmix and its gorgeous array of sounds can also be used as the audio engine for both iPhone and Android apps.

With its versatility and portability, RTcmix exists as a powerful program designed for composing highly customizable computer music in any of a variety of platforms. At its heart lies the work of Paul Lansky's MIX and CMIX programs and to an extent Max Mathews' MUSIC-N languages, thus giving RTcmix a branch on the MUSIC-N tree shared by CSound and SuperCollider.

Because of its long and rich history of flexibility, there is no reason to suspect that RTcmix will suddenly become irrelevant as the next-greatest-iDevice or platform hits the market. In fact, I'm already imagining a world where I can be eating dinner and coding RTcmix scores using my retinas and stealth mind control in my Apple or Google contact lenses...

|| Sonata I: Downloading and Installing RTcmix ||

An ongoing theme in this book is the notion that RTcmix exists in various guises and that this flexibility is part of its appeal. The heart of RTcmix lies in its score files, or scripts. The suite of instruments, commands, and the anatomy of the score file will remain the same no matter how you decide to use the program.

There are two main ways to interact with RTcmix. The first is from within the Mac OS Terminal Application, which involves a very basic knowledge of UNIX, but is a very rewarding experience for the user. The second is from within Max or Pure Data, each a powerful graphical programming software suite of tools.

After installing RTcmix on the command line, we'll look another Application, CMixRun, which integrates RTcmix into a welcoming user experience.

Our Macintosh Operating System has a wonderful set of tools to streamline the process of downloading and installing software to your computer and Apple currently boasts an App Store which serves as a one-stop-shop for all of your software needs, requiring little more than a simple point-and-click to initiate and execute your download. Because RTcmix isn't proprietary software (meaning that you don't have to pay for it) and is open source (meaning that you'll be able to alter the program itself in any way you wish), we won't find it in the App Store. Instead, we will have to download the program from its website and execute the installation manually, which is more fun, anyway.

We will be running our installation of RTcmix from the command line using UNIX. This is a powerful operating system that is itself integral to the Mac OS. In fact, using UNIX is a central part of our work with RTcmix and we'll be exploring and utilizing UNIX with a utility on your Mac called Terminal, which can be found at /Users/User/Applications/Utilities/Terminal.app.

Here's an example of the default Terminal window, which is white with black lettering, though you can change your window's appearance in Terminal's preferences.

My Terminal Utility is in the directory /Users/jerod_s/Applications/Utilities/Terminal.app. We'll be getting around on our Mac using UNIX and Terminal in a bit

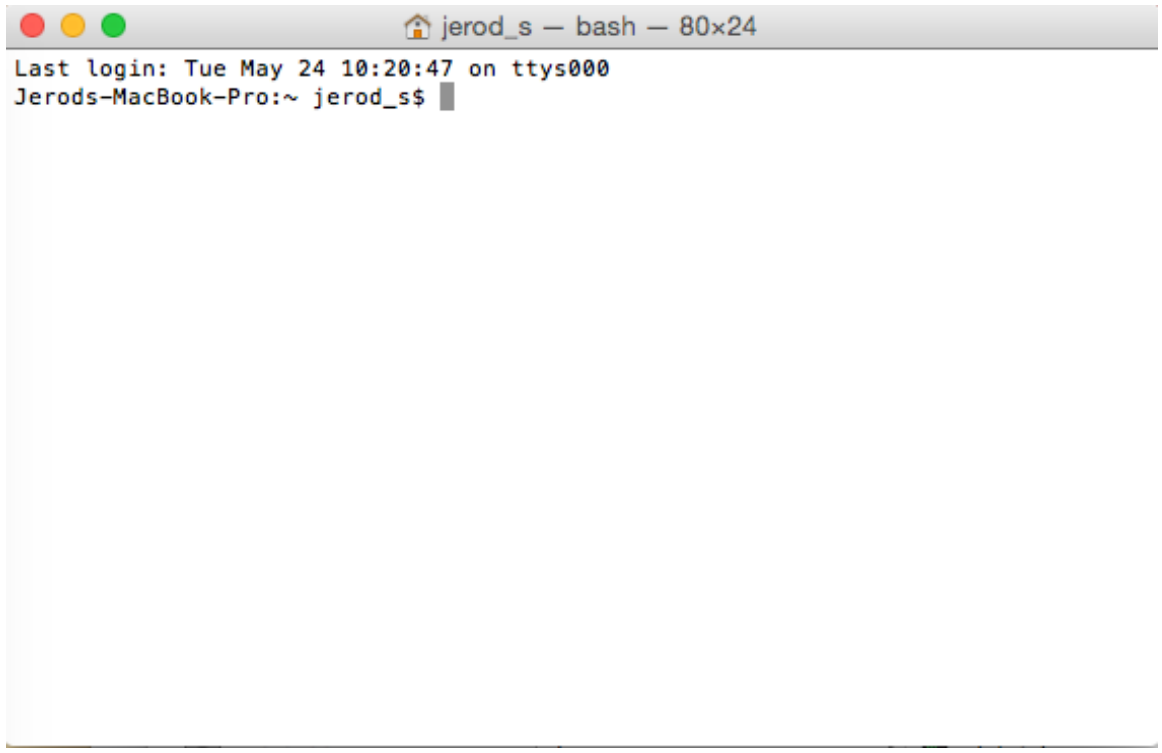

Figure 1: Default Terminal Window

We'll also be compiling RTcmix manually, so there are a few steps we'll need to complete in order to get us started. To wit:

1. We'll install the GNU compilers for OS X.

2. We'll download the RTcmix package from the main RTcmix website.

3. We'll install RTcmix from the Terminal.

The Developer Tools Library — which contains the GNU compilers — is a suite of tools found within XCode, a major software program from Apple that affords the user the ability to design software applications for their Mac or iOS device. If you're interested, you are more than welcome to head to the aforementioned App Store and download XCode, which at 2GB is a pretty beefy application. Otherwise, type the following into your Terminal window.

```
gcc
```

If you don't already have the GNU compilers or the Developer Tools Library installed — which I suspect many of you reading this won't — simply follow the prompts on the screen and they will be installed for you. What exactly is this gcc command and why did we need it?

The instrument libraries, scorefile functions, and other intrinsic components that make up the RTcmix program are all written in the C and C++ computer programming languages. Because RTcmix is distributed freely and open-source, you (the user) will be able to take a look at all of this code that went into designing the program itself. However, C and C++ are examples of so-called high-level programming languages, meaning they borrow terms and syntax from our spoken language in order to help create a somewhat intuitive programming experience.

As mentioned in the Prelude, RTcmix scripts are written the high-level languages MINC, (MINC-is-not-C), Perl, or Python. Once we delve into the details that constitute the anatomy of an RTcmix scorefile, we'll be able to better differ-

entiate MINC and C, as well as MINC and Python, but for now it's important to understand their relationship to low-level languages.

Also referred to as assembly-language or machine code, low-level languages correspond to and interact closely with the computer's unique hardware architecture. An incredibly detailed sequence of instructions, these assembly-languages get at the heart of the machine: Streams of binary data. Represented as bits 0 and 1, these strings of numbers assist with complex tasks that your computer executes each time you put it to use. However, because of the level of detail and level of foreignness to our written and spoken languages, it is useful for us to have a translator, which we refer to as the compiler.

The name GCC is short for GNU Compiler Collection. This GCC compiler serves as the bridge between the code that built RTcmix (C and C++) and the machine-language that supports and executes commands in the Macintosh OS. So, in order to install RTcmix on our OS, we need these GCC compilers to help our machine understand what it is that RTcmix will be asking of it. And now that we have these compilers, we can continue installing RTcmix.

Type command-t to open a new tab in the Terminal window, and type in the following line of code.

```
cd /Applications
```

Once you hit enter, your Terminal should look like something akin to the following three lines.

> You'll note that your Terminal window will keep you updated on RTcmix's download progress as you await its arrival.

```
Last login: Fri Aug 30 11:34:10 on ttys000
```

```
localhost:~ mycomputer$ cd/Applications
localhost:Applications mycomputer$
```

We just changed our directory using the command cd into Applications, which you're already familiar with on your computer. Now our Terminal is working from the Applications directory, which is where we want to put RTcmix once we've downloaded it. Again, this process won't be as familiar to you as is getting a new App from the App Store, but hang in there and type the following into your Terminal window.

> To be clear, it is a dot followed by a backslash and configure, all in one string of characters. Then, one space, followed by two dashes and with. Then another space

```
git clone https://github.com/rtcmix/rtcmix
```

It's of course important to type this in exactly as I wrote it above, with the correct spaces. But so what exactly did we do here? RTcmix's source code lives on a website called GitHub, which allows communities of users the ability to alter and collectively improve and expand upon curated software projects and packages. While we could've navigated to RTcmix's GitHub website and downloaded everything from there and placed it into our Downloads folder, this is a much more streamlined way that hopefully gets you feeling more and more comfortable interacting with your Terminal and UNIX. Specifically, we've cloned the RTcmix source code from GitHub using the git command, which is pretty easy!

Now that we've saved RTcmix into Applications, we need to change directories into the new RTcmix folder to continue installing and building it.

```
cd /RTcmix
```

At this point, you are now working inside the RTcmix folder and have access to all of the items within it. This is important, because now we're going to use those GCC compilers to build the program. To get that process going, type in the following line, making sure that you write it exactly as written below so that we can configure our build with Python.

```
./configure —-with -python
```

After you hit enter, your Terminal window should be scrolling through line after line after line of syntax, which details the play-by-play of the build process. This is a look at the internal workings of the build process that many don't ever see when installing software from a Setup Wizard or other GUI download assistant. Depending on the speed of your machine, this might take a minute, so hang tight.

Once the process is complete, your Terminal readout should look something like this, which is RTcmix's way of letting you know that the process was successful and complete.

```
Summary...
   Building 64-bit binaries
   Build with multi-threaded support.................. : false
   Build with Perl support............................ : false
   Build with Python support.......................... : true
   Build with ALSA support............................ : false
   Build with JACK support............................ : false
   Build with NetPlay support.........................: false
   Build with FFTW support............................ : false
```

```
Build with OSC support.......................... : false
```

Of special note of course is the fact that we've built RTcmix with Python. You are of course able to select as many options as you'd like, though you'll also need some ancillary downloads to get those options to work. For now, there is no need to to configure with anything else, thus allowing us to make music right away using the standard MINC or Python languages.

Now that RTcmix has been configured, we need to "make" all of the instruments and scorefile commands in its library. This is a simple process that only involves typing

```
make
```

and hitting enter. This process should take longer than configuration, so be patient and let it do its thing. Once that process is done, the last step is to actually install those instruments and commands onto your computer, by installing it.

```
make install
```

RTcmix is now installed on your machine! We are interested in an executable file found in RTcmix's /bin folder called "cmix." To verify that the entire installation process worked correctly, change directories to your bin and enter in the following line of code.

```
./cmix
```

> Hopefully you're comfortable at this point with changing directories on your computer. If not, remember to

You'll know that RTcmix successfully installed if your Terminal prints out the following line.

```
--------> RTcmix 4.2.0 (cmix) <--------
```

All of our installation work went into ensuring that this executable command is working properly, which allows us to have our score files interpreted by our computer and realized as actual sounds. To verify that the Python build worked as well, simply type in "pycmix" in place of "cmix" above. Type control-c to interrupt the executable and return to your working Terminal window.

As of now, we're only really able to interact with the pycmix or cmix commands while working in the bin folder and that isn't the most convenient way for us to work. We want our computer to be able to find and use that command no matter what directory we happen to be in, so we need to point it in the right direction.

Type in the following line of code to your Terminal.

```
open ~/.bash_profile
```

This will open a file called .bash_profile in your TextEdit application. Note that there is nothing in the file itself, but we are going to add the path to these pycmix and cmix commands so that our computer can locate them. So, in the .bash_profile file, add the following paths.

```
export PATH=$PATH:/Users/yourcomputer/Applications/RTcmix/bin/cmix
export PATH=$PATH:/Users/yourcomputer/Applications/RTcmix/bin/pycmix
```

Again, my computer is named "jerod_s" and that is what goes in place of the "yourcomputer" location above on my machine, so of course yours will be named differently. What we are doing here is adding a new path to your bash shell's profile, or giving our computer the opportunity to find specific executables in addition to all of its intrinsic commands like cd, and sudo, and cal, and others.

Go ahead and save that file (command-s) and quit TextEdit (command-q) and return to your Terminal window. At this point, we've only temporarily set the new paths, but of course we want to draw upon them after we reboot our computer or quit and restart the Terminal itself.

```
source ~/.bash_profile
```

This command ensures that the new paths that we created are now built into our profile and can be used at any time and from any location. Let's test this in two ways.

```
cd
```

Typing in cd alone will get you back to your home directory, so now we're out of RTcmix's bin folder.

```
cmix
```

If all went well, you should be able to type in the pycmix or cmix command and see the same line that we saw earlier when working in the bin folder.

```
--------> RTcmix 4.2.0 (cmix) <--------
```

Type control-c to interrupt the process and return to a Terminal window that's ready to start playing some score files!

> Admittedly, there were a lot of steps involved in getting the pycmix and cmix commands ready for use on your

RTcmix comes with a variety of help files, useful documentation, and examples to draw from, which you'll find in the RTcmix folder. More specifically, they can be found in the sample_scores subfolder, so let's navigate ourselves there to start playing some sounds.

```
cd /Applications/RTcmix/docs/sample_scores
```

Once you are there, let's get a list of all of the files that are contained within.

```
ls
```

You should see a long, long list of files that end in .sco, .pl, and .py, which denote MINC, Perl, and Python scripts, respectively. Let's take a look at the STRUM1.sco example in our Terminal window.

```
nano strum1.sco
```

What is this madness?! Could that be a full-blown text editor right here in the Terminal? You bet!

This is what an RTcmix script looks like, which in this case is written using MINC. You'll see some comments from its author Brad Garton, as well as your first introduction to the powerful STRUM( ) instrument. Type control-x to exit the nano editor and return to your Terminal. Now we'll play the score file itself.

```
cmix < strum1.sco
```

Very cool sounds, yes? When the script has finished playing, your Terminal readout should contain the following text toward the end of your window.

```
random:
============================
sampfunc:  2 4.98959
============================
random:
========<rt-queueing>=======
START:  15 1 7.1 1 0.1 10000 1 0.812958
START:  Setting phrase curve to all 1's.

closing...
Output duration: 16.02 seconds

Peak amplitudes of output:
  channel 0:  8421.648438 (-11.80 dBFS) at frame 406057 (9.20764 seconds)
  channel 1:  8106.027344 (-12.13 dBFS) at frame 569375 (12.911 seconds)


Jerods-MacBook-Pro:sample_scores jerod_s$
```

There are a number of useful things that've been printed out for you and we'll get to them in due time. For now, make note that the duration of this score file was 16.02 seconds and that we received audio output on channels 0 and 1.

When executing a script, the "<" key acts as a pointer to the score file that you'd like to play. Let's say that for example you decided to open this script in TextEdit instead of nano. Your TextEdit window should have an icon with the .sco file's name seated above the text window with all of the MINC code. By clicking on that icon and holding it down, you'll be able to move it across your screen, much like you would do if you wanted to open a file with an application found in your dock. Then, you can drop it into your Terminal window after the "<" sign. This is a shortcut to tell RTcmix (or any application on your computer, really) in which directory to find the file that you are attempting to open.

For example, if I wanted to open John Gibson's simplesynth.py file — which uses Python — I could open it in TextEdit, drag that top icon into Terminal, and see the following in my window:

> I can't help but introduce TextWrangler here, from Bare Bones Software. You can find the download link at

```
pycmix < /Users/jerod_s/Applications/RTcmix/docs/sample_scores/simplesynth.py
```

Provided that your Terminal reads something similar, you can type "enter" and RTcmix will play the script!

If everything up to this point has worked smoothly and your computer is executing cmix or pycmix and making wonderful sounds, then we've completed the installation process, which is great, but let's get down to finally making our own sounds.

We're going to make our own from scratch and briefly discuss each step in creating our own score files, knowing that another sonata in this book is dedicated to a more detailed explanation of an RTcmix script's constituent parts.

Change directories to your Desktop and get back to your nano editor by typing, well, nano, and enter the following code line for line to start a new script.

```
rtsetparams(44100, 2)
load("WAVETABLE")
```

Script is to score file as score file is to script. Apologies in advance for the interchangeability of these two con-

Thus far, we've accomplished three critical tasks. First, we told RTcmix that we'll be working with the standard, CD-quality audio sampling rate of 44,100 samples/second or 44.1 kHz and we've also stated that we'll be working with two channels of audio output. This is the *raison d'être* of rtsetparams( ), which stands for "real time set parameters." Later on, we'll be tweaking a few more things using rtsetparams( ), but please note that this will be the standard header on every score file that you'll write.

We've also loaded the WAVETABLE( ) instrument, which is a powerful wavetable oscillator that we'll encounter several times in this book. If you didn't include the quotations in load("WAVETABLE") or perhaps didn't include the capitalization, you're going to notice that your script won't work, so it's important to get in the habit of understanding the proper syntax. RTcmix interprets text very literally, and all of the quotes, caps, commas, and parentheses need to be exact in order to run. After setting the parameters for your script, your next step will always be to load the instruments that you'd like to use, so this whole load( ) business will become second hand in no time.

Hit "return" twice to give yourself a double space and type in the following:

```
WAVETABLE(0, 5, 20000, 440, 0.5)
```

Each time you call on WAVETABLE( ) to execute a command, you need to tell it what to do by specifying its parameter fields or "p-fields." In essence, you've told it to start making sound right away (0), produce a sound for five seconds (5), at a given amplitude (20000), with a frequency in cycles/second (440), directly in the middle of the stereo field (0.5). Again, we'll get into much more detail in the next chapter, but for now, we just want to hear some sound!

```
rtsetparams(44100, 2)
load("WAVETABLE")

WAVETABLE(0, 5, 20000, 440, 0.5)
```

All of the example score files are included online for downloading from my personal website, though it's

Score file 1: Simple sine wave

To save your script, type control-o and give your score file a name. Something like firstscript.sco will work. Hit "return" and then control-x to exit nano.

```
cmix < firstscript.sco
```

Well hey there! Congratulations on your first sound in RTcmix! Such beauty! If you'd like to save this watershed moment of digital music and show it off to your family at Thanksgiving, you might want to create a folder on your Desktop to save all of your score files into, so that you can quickly find them later on.

There are a myriad of other ways to interact with RTcmix and we've gone through the primary method of interacting with the sounds and commands contained within. Of special note of course is the fact that we don't have a graphical interface, point-and-click methodology or any other more intuitive components that we might expect in, say, a big-box digital audio workstation or other software program that we might be paying for. For those familiar with Max and Pure Data, the [rtcmix~] object is available, which we will look at toward the end of this book.

> Later on, we'll explore a variety of ancillary downloads to further enhance your RTcmix experience - such as

All of this isn't to say, however, that others in the RTcmix community haven't already contributed new ways of using RTcmix and we'll look at one, CMixRun, which is my favorite and we'll find is very feature-rich.

Neil Thornock is the author of CMixRun, a front-end application that includes all of RTcmix in another easy to use graphical user interface. You can download the application from his site on Sourceforge: http://sourceforge.net/projects/cmixrun/files/?source=navbar and navigate to the link for his "CMixRun.pkg" file, which contains everything that you'll need.

Once you've downloaded it, double click on the icon to begin the installation process. This will automatically place the CMixRun application into your Applications folder.

When the program opens, you'll find a really useful interface with a series of windows.

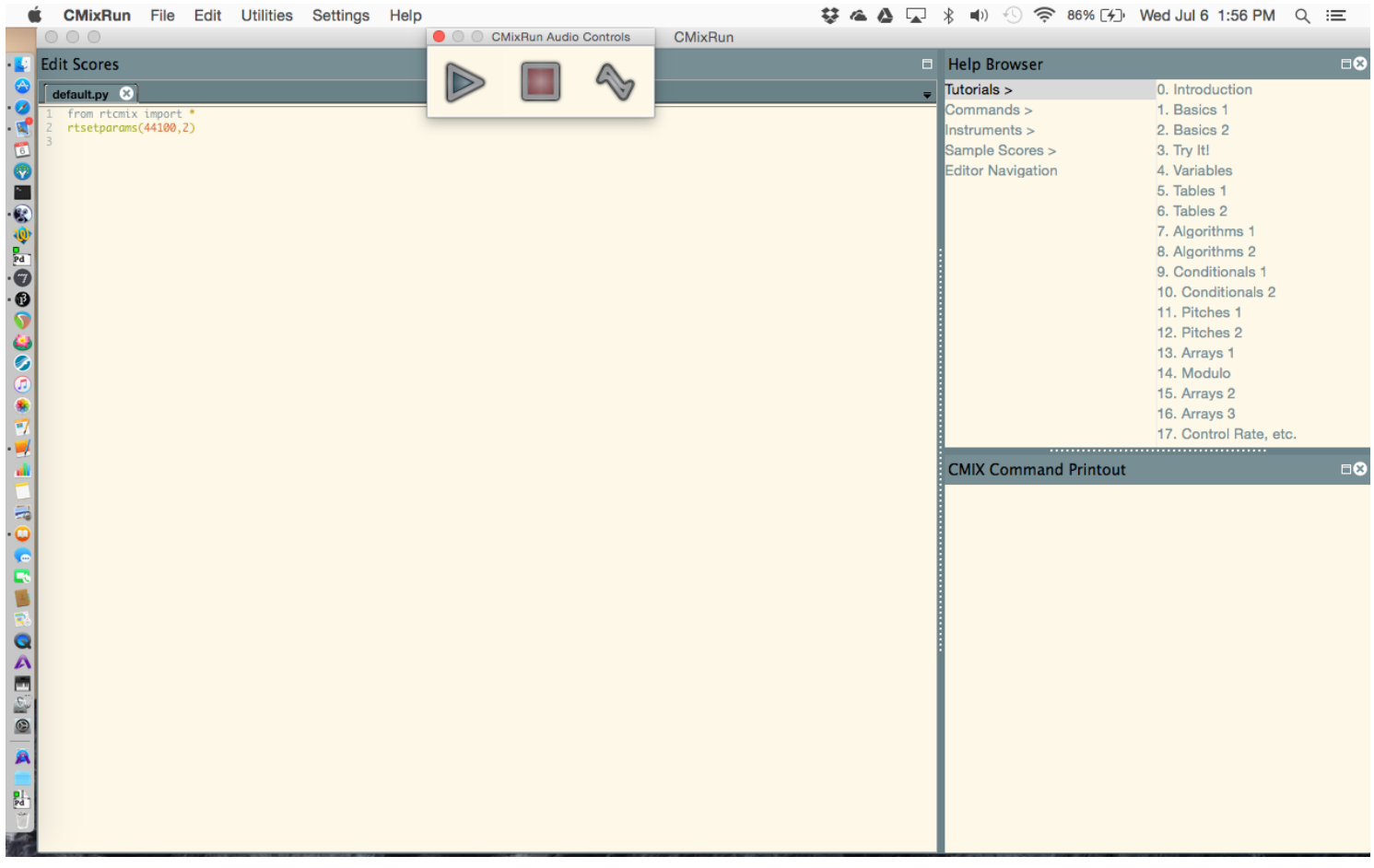

Figure 2: CMixRun main window

Your main window will serve as your scripting window and includes line numbers and text highlighting to better view the commands and instrument calls that we'll be utilizing in our work and is compatible with either MINC or Python. To the upper right, you'll find a series of tutorials written by Neil, which I highly encourage you to scour while working on this text. The window to the lower right will let you know if your score files have run successfully or if they have any errors. The floating playback controls will play, stop, and save your current script.

Go ahead and double-click on "0. Introduction" and the script will open in your main window. Follow Neil's instructions and you'll hear one of his score files in action!

Sonata I: WAVETABLE( ) Sounds

Using your preferred method of generating sounds with RTcmix, create a brief series of sounds that uses several instances of WAVETABLE( ) with varying start times, durations, amplitudes (keep these in the range of 5000 to 20000) frequencies, and pan locations (in the range of 0 to 1). You might try to generate chords or a simple melody with or without accompaniment.

|| Interlude I: Basic UNIX for the RTcmixer ||

Each time we sit down to work on an RTcmix score, we'll first head to our Applications folder or Dock, search for Utilities, and open Terminal. This small, unassuming window allows you to type in a variety of commands and have them executed on your operating system, hence why it is often referred to as the "command line."

The purpose of this interlude is not to explore every feature of the Terminal at our disposal, but to get a quick feel for the types of commands that you'll be using each time you use RTcmix. While going over all of the intricate details and walking the entire labyrinth of UNIX isn't exactly our goal here either, getting some useful, basic information from UNIX will greatly enhance not only our experience with RTcmix, but also our experience in working with our computers.

At the heart of the Macintosh Operating System (OS X) lies UNIX, a capable operating system that often goes unnoticed in your day-to-day computer business, because Apple has lovingly created OS X with a gorgeous user interface that we are of course so accustomed to using already. For example, if you were to create a new folder on your Desktop (by going to your top menu and selecting File --> New Folder) you might not be aware that it is possible to accomplish the same task in Terminal, by typing the command mkdir.

You are probably familiar with Apple's native calendar application, Calendar. UNIX has its own, albeit spartan, calendar and can be accessed by using the following command.

```
cal
```

Your window should give you a view of the calendar for the current month. You can also check to see what the calendar looked like for the month and year that you were born. For example, here's what the calendar looked like in my birth month and year.

```
cal -m -y 12 1982
```

In order to change the view of our calendar, we had to use a few so-called flags. In this case, we wanted to view a particular month in the future (or past), so we needed to let the Terminal know that we would be specifying the month, as well as the year, with the flags -m and -y, making sure to include a space between each individual flag. After that, we wrote, in order, the month and year in question. For some, the configuration process while installing RTcmix was done with flags, especially if you chose or will be choosing to write your score files in the Python programming language, in which case you wrote ./configure --with-python.

Each UNIX command (like cal) contains its own help file, which is a great way to learn more about the processes that you are executing. For example, type the following into your Terminal and read the corresponding printout.

```
man cd
```

You just prompted your Terminal to give you some more information about the command that allows you to change directories, cd. When you are done reading, type the letter "q" and you'll exit the help menu. As mentioned in the section on downloading and installing RTcmix, it's important to remember that as you navigate around your computer — moving from your Applications to Documents to your Downloads and back — you could move in the same way using cd. Now if you installed RTcmix into your Applications folder, take a moment to practice navigating your way there.

> Now, you might need to add a slash in front of applications, so you'll be typing cd /Applications for this par-

It might help to type pwd in order to print your working directory, or see where you are currently working. Calling cd all by itself will send you back to your home directory and if you want to be super stealth, try typing two periods like this

```
cd ..
```

and after doing another pwd you'll notice that you'll find yourself up one level, back where you came from.

While it's all well and good to navigate around directories, we need some commands that will be useful for us in conjunction with RTcmix. For example, we can list the files within a directory.

```
ls
```

You should see a number of items listed in your Terminal window, which should look something like this:

```
Macintosh:~ jerod_s$ cd /Applications/RTcmix/
Macintosh:RTcmix jerod_s$ ls
AUTHORS      config.h.in      lib
ChangeLog    config.log       liblo-0.27
INSTALL      config.status    makefile.conf
LICENSE      config.sub       makefile.conf.in
Makefile     configure        pkg
NEWS         configure.ac     pkg-config
README       defs.conf        scores
THANKS       defs.conf.in shlib
aclocal.m4   docs             site.conf
apps         genlib           snd
bin          include          src
config.guess install-sh       test
config.h     insts            utils
```

We've listed all of the items in our directory and for now many of them might be confusing. You can probably guess what the AUTHORS or NEWS, or README, or THANKS files are all about, but there are certainly a few listed in here that while now seem at first blush vague and perhaps cryptic, are going to be of interest to us, to wit:

> If you're curious, go ahead and type in
>
> open THANKS

1. /docs contains all of the sample scores and sample code for RTcmix

2. /lib houses the source code for RTcmix's instruments, written by its authors

> For those who plan to alter/change the program in some way, this will be a place for you to check out in more

3. /snd contains sound files for you to sample and please listen to Looch

4. /bin has the CMIX executable, as well as PYCMIX for Python, etc.

5. /src keeps all of RTcmix's source files and source code

Take a moment to navigate to /docs/sample_scores and type in an ls command to view all of the sample scores that are available to you right out of the box. Each one of these scores are there for you to alter, explore, and utilize in creating your own music. Moreover, each one was created by one of RTcmix's authors, which is a great way for you to see how they write their code. As you move along in the program, you'll find syntactical tendencies with your own coding and will quickly notice that despite the program itself not really changing in any way, many different users will have very different approaches to coding their music in RTcmix.

Using cd .., find your way back to your home folder for RTcmix and navigate to /insts. When you list the contents, you should see something like this on your screen:

```
Makefile    bgg     joel    std
stk         base    jg      maxmsp  std-04
vccm
```

Within this folder are more example score files and documentation for RTcmix, straight from the authors who created each respective instrument. For example, you can navigate to John Gibson's instrument folder, Brad Garton's, Joel Matthys', or the standard RTcmix instruments, those in the synthesis tool kit, or those that can be used in conjunction with Max (which we'll cover a bit later.)

As we move along and create many, many new score files, we'll definitely want to be able to search for certain commands that we've previously used so that we can recall them later. For example, let's say that we are trying to make some sounds using granular synthesis, and we want to use a previously coded Hanning window for our grain envelope. Although we know we used it, we can't quite remember the exact score file that it was utilized within. This is where some UNIX search tools will come in handy, including the powerful grep command.

Navigate your way into the /RTcmix/docs/sample_scores directory and type the following into your Terminal window:

```
grep hanning *
```

and you'll see a readout that looks like this:

```
Macintosh:sample_scores jerod_s$ grep hanning *
CONVOLVE1_2.sco:src_env = maketable("window", 1000, "hanning")
CONVOLVE1_2.sco:window = maketable("window", 1000, "hanning")
GRANSYNTH1.sco:envtab = maketable("window", 2000, "hanning")
GRANSYNTH2.sco:granenv = maketable("window", 2000, "hanning")
GRANSYNTH3.sco:envtab = maketable("window", 2000, "hanning")
GRANULATE1.sco:envtab = maketable("window", 1000, "hanning")
GRANULATE2.sco:envtab = maketable("window", 1000, "hanning")
GRANULATE3.sco:envtab = maketable("window", 1000, "hanning")
GRANULATE4.sco:envtab = maketable("window", 1000, "hanning")
JCHOR1.sco:grainenv = maketable("window", 1000, "hanning")
JCHOR2.sco:grainenv = maketable("window", 1000, "hanning")
```

```
JCHOR_8chan.sco:grainenv = maketable("window", 1000, "hanning")
JGRAN1.sco:genv = maketable("window", 1000, "hanning")
JGRAN2.sco:genv = maketable("window", 10000, "hanning")
JGRAN3.sco:genv = maketable("window", 10000, "hanning")
JGRAN4.sco:genv = maketable("window", 1000, "hanning")
JGRAN_FLANGE_REVERBIT.sco:genv = maketable("window", 10000, "hanning")
JGRAN_JDELAY.sco:genv = maketable("window", 10000, "hanning")
JGRAN_REVERBIT.sco:genv = maketable("window", 10000, "hanning")
SHAPE1.sco:indexguide = maketable("window", 1000, "hanning")    // bell curve
STEREO3.sco:env = maketable("window", 10000, "hanning")
grep: disk-based: Is a directory
```

We've just queried our directory to seek out and return all of the instances where the string "hanning" is located. So, within all of the score files, it managed to find quite a few instances of the Hanning window! Looking at the first returned line, we can see that the string "hanning" was used in the CONVOLVE1_2.sco score file, where it was used to declare the variable src_env (presumably the audio "source" envelope shape in convolution) and was utilized in a maketable command (which we'll use A LOT) and its requisite variables ("window", 1000, "hanning"). You're probably wondering why we used the * character for our search: This is a so-called "wild card" that can be used in the grep command. In this case, we're using our wild card to search only in the /sample_scores directory.

> Another fun tip is to use the up arrow to repeat your last command. This is really helpful for quickly re-

Your Terminal window might be a bit messy right now, so for the neat and tidy, type

```
clear
```

which will clear the contents of your window. Well, we haven't actually cleared the contents of the window per se. It's all still there and you can find it by just scrolling up or typing the up arrow.

While the world of UNIX is in and of itself varied and exciting and well worth exploring in more detail, we're now equipped with most of the basics that we'll be using in conjunction with our work in RTcmix. Here and there we might come across a few new UNIX commands, but with this Interlude, we should be ready to freely navigate our computers and make some music!

Interlude I: Scavenger hunt

Using the Terminal and your new UNIX wizardry, locate and open any of the sample score files for the STEREO( ) instrument found in /path/to/RTcmix/docs/sample_scores. Then, rewrite at least two of those score files, verbatim, then freely manipulate them into a new 30" piece.

|| Sonata II: The Anatomy of an RTcmix Score File ||

While certainly daunting at first blush to look at a score file that contains hundreds of lines of code, each RTcmix script begins in exactly the same way, calls instruments just like any other, and sets up routines for your computer to execute. We're going to start small and bit by bit construct something fun, all the while gaining a good understanding of what exactly goes into an RTcmix score file.

As mentioned earlier, each script will intrinsically begin by setting up our audio sound card and loading instruments.

Start a new RTcmix script and type in the following text:

```
rtsetparams(44100, 2)
load("WAVETABLE")
```

Again, we are setting our sampling rate to 44,100 samples per second and utilizing two channels of audio output. If, say, we have a super fancy new Fire-Wire sound card that we'd like to try out and we know that it can sample at much higher rates, we could declare 96,000 or something even higher, as long as that sampling rate is supported. Moreover, let's say we're working in a studio that boasts eight channel surround sound, in which case we could opt to declare eight channels of output.

One feature that we will start using now and is highly, highly recommended for your work is the comment. By placing two slashes in front of any text, RTcmix will leave that bit out during execution. So, I can comment the code we've written to help me remember what each line does. If you have lots of text that you'd like to use as a comment, you can use block comments instead.

```
rtsetparams(44100, 2)  //set parameters, rtsetparams(samplerate, channels)
load("WAVETABLE")      //load the WAVETABLE instrument

/*
A slash and then a star will make a block comment.
Anything inside of those characters

will be

commented out.

Use a star then slash to finish the block comment.
*/
```

Were we to omit those slashes after rtsetparams( ), we'd get an error in our Terminal readout, which looks something like this:

```
Macintosh:~ jerod_s$ cmix < /Users/jerod_s/Desktop/mygreatfile.sco
--------> RTcmix 4.0.1 (cmix) <--------
============================
rtsetparams:  44100 2
Audio set:  44100 sampling rate, 2 channels

*** ERROR [parser-yyerror]: near line 1: syntax error
```

Looking back, we'd find our way to line 1 and make note of our error, in this case the fact that we forgot to include our slashes for the comments.

WAVETABLE( ) is a great instrument to use when starting out with RTcmix, as it is able to synthesize all of the standard waveforms (sine, sawtooth, triangle, square) and when used in conjunction with maketable( ), gives the user the ability to create custom waveforms, subtract and add respective partials and adjust the amplitudes of those partials, or even make some fun, grainy speaker clicks to use as textural material. In short, it is a highly versatile instrument and one that we will utilize often while working through the program.

Earlier, we tested our installations of RTcmix by implementing a simple sine wave at 500 Hz. Let's rehash that, but keep it a bit more bare bones for now.

```
rtsetparams(44100, 2)  //set parameters, rtsetparams(samplerate, channels)
load("WAVETABLE")      //load the WAVETABLE instrument

WAVETABLE(0, 5, 20000, 500, 0.5) //play a sine wave for 5 seconds
```

Score file 2: Simple sine wave with comments

At this point, it is useful to introduce the notion of variables and how they can be used in RTcmix, because as we can already see, housed within the WAVETABLE( ) parameters are a series of values separated by commas that for now are somewhat meaningless to us. A variable is nothing more than a declaration inside of your script. For those who are more familiar with C programming, we don't need to specify whether our variable is an integer or a float or anything else, which is quite handy in the MINC parser. As a simple example, I can create variables for integers to print to the Terminal window by typing the following code:

```
x = 2
y = 3

sum = x + y
print(sum)
```

Score file 3: Print to Terminal window using variables

In the case of our example using WAVETABLE( ), it is going to be useful to substitute variables in place of our integer values so that we better understand how they function.

As we saw earlier, each RTcmix instrument contains a series of parameter fields, or p-fields, which need to be declared in order to properly execute sound. They are listed in order within the parentheses and are separated by commas. In this case, WAVETABLE( )'s p-fields are determine the following parameters:

p0 = start time

p1 = duration, in seconds

p2 = amplitude, or loudness

p3 = frequency

p4 = placement of sound in the stereo field (optional p-field command)

p5 = waveform specification (also optional)

As we will encounter later on, many of the p-fields for RTcmix's instruments can be updated in real-time, or

If you're like me, you probably find it helpful to remember what these p-field commands are, not necessarily by memorizing them, but by putting names or vari-

able names to them within your script. We can do this within the parentheses that call the instrument, like so:

```
WAVETABLE(start = 0, duration = 5, amplitude = 20000, frequency = 500, pan = 0.5)
```

> Here's a critical note: Localizing variables within the call to each RTcmix instrument is not possible in Python.

However, its not always going to be the case that we want to localize those variables so specifically. Once we add a number of other instruments and effects to our scripts, we might want a global start time, or a duration that works for a variety of different sounds. So, here is a nice looking script that first specifies our p-field parameters as variables and then puts those variables into the call to WAVETABLE( ).

```
rtsetparams(44100, 2)
load("WAVETABLE")

start = 0
duration = 5
amplitude = 20000
frequency = 500
pan = 0.5

WAVETABLE(start, duration, amplitude, frequency, pan)
```

Score file 4: Sine wave with variables

You'll notice that p4, panning, is one optional parameter for you to specify. Each RTcmix instrument that you encounter will have some p-fields that are absolutely necessary for their function and some p-fields that are optional. If we

didn't specify a pan value (0.5 in the middle, 0 to the right, 1 to the left), RTcmix would send a default 0.5. Likewise, p5 is optional and defaults to a sine wave. If we want to utilize one of our other available waveforms, we need to introduce the maketable( ) command. Try typing this after the declaring the variable for pan

```
waveform = maketable("wave", 1000, "saw")
```

and include it in your WAVETABLE( ) parameters. You should have a line that now looks like this,

```
WAVETABLE(start, duration, amplitude, frequency, pan, waveform)
```

and sounds the same 500 Hz wave as before, but instead of a sine wave, now plays a sawtooth wave. Each time that you call on maketable( ), you need to specify a few parameters. First, you need to indicate the type of table that you need and put it in quotations. Since we want to draw distinct waveforms, we need to write "wave." The 1000 indicates the number of points on the x-axis of the table that we are drawing.

1000 is a nice, easy number to remember and is a good, standard table size. While a more detailed explanation of the various types of maketable( ) will follow in the next interlude, note for now that "wave" can provide you with the following waveforms:

"sine" -- sine wave
"saw" -- sawtooth wave

"sawX" -- a sawtooth wave with X number of harmonics

"sawdown" -- sawtooth wave whose first curve moves downward

"sawup" -- sawtooth wave that goes upward

"square" -- square wave

"squareX" -- square wave with X harmonics

"tri" -- triangle wave

"triX" -- triangle wave with X harmonics

"buzz" -- pulse wave

"buzzX" -- pulse wave with X harmonics

Thus far, we've approached our score file from a fairly methodological angle. Our goal was to make a simple sine wave, have it sound for a few seconds, then turn off. What if, however, we wanted to start constructing scales and modes and impressing our friends with rising whole-tone-scale-dream-sequence sounds? For that, we can call a large array of WAVETABLE( ) instruments:

```
rtsetparams(44100, 2)
load("WAVETABLE")

start = 0
duration = 5
amplitude = 10000
pan = 0.5
waveform = maketable("wave", 1000, "sine")
WAVETABLE(start,   duration, amplitude, cpslet("C4"),   pan, waveform)
WAVETABLE(start+1, duration, amplitude, cpslet("D4"),   pan, waveform)
WAVETABLE(start+2, duration, amplitude, cpslet("E4"),   pan, waveform)
WAVETABLE(start+3, duration, amplitude, cpslet("F#4"),  pan, waveform)
WAVETABLE(start+4, duration, amplitude, cpslet("G#4"),  pan, waveform)
WAVETABLE(start+5, duration, amplitude, cpslet("A#4"),  pan, waveform)
WAVETABLE(start+6, duration, amplitude, cpslet("C5"),   pan, waveform)
```

How in the world did we get to use letter names in this script? Well, RTcmix doesn't always need to repre-

```
WAVETABLE(start+7, duration, amplitude, cpslet(“D5”),   pan, waveform)
```

Score file 5: Whole tone scale

Typing this out in full and not taking advantage of at least copy and paste can get pretty maddening pretty quickly. The snippet of code above does in fact work: With a new duration of one second, it plays C4, then after one second plays D4, then after another second E4, and on and on and on. There really isn't any limit to the number of notes that you can play in sequence, but my hope is that its evident that we're doing more work than necessary. We're on a computer after all, and computers are designed to make routine tasks easier for us. For this particular situation, we'll need to construct a loop.

Loops give us the ability to repeat a number of tasks that we designate for however many iterations we desire. These loops have a pretty detailed construction that we'll utilize quite a bit over the course of our work, so its best to just take the leap of faith now, explain the concept, and commit it to memory.

```
rtsetparams(44100, 2)
load(“WAVETABLE”)

start = 0
duration = 1
amplitude = 10000
note = 60  //the MIDI note number for middle C, or C4
pan = 0.5
waveform = maketable(“wave”, 1000, “tri”)

for(start = 0; start < 7; start = start + 1){
     WAVETABLE(start, duration, amplitude, cpsmidi(note), pan, waveform)
```

Here is yet another way to designate pitch. MIDI note numbers are a great way to represent pitch and cpsmidi( )

```
note = note + 2  //increment by a whole step
}
```

Score file 6: Whole tone scale from a loop

Okay so here's what we just accomplished. Our loop construction is literally interpreting us saying, "Make the variable 'start' equal zero. Then, as long as start is less than seven, increment start by one."

That is, any commands and text that are placed within the curly braces { }. The indentation is a matter of style

You'll notice that because we also utilized "start" in our call to WAVETABLE( ), our first start will equal zero, then one, then two, then three..., just like it did above when we had to physically write it over and over. Way easier.

Readers who are super familiar with the C programming language are probably wondering why our incre-

Moreover, starting a loop will give us the chance to change variables within the loop structure, as we did when we wrote that we wanted "note" to equal itself, but add two. Thus, our variable "note" will first equal 60, then 62 (60+2), then 64 (62+2), then 66 (64+2), etc.

We're incrementing by one each time through the loop and all of our durations are also equal to one. Thus, we're getting fairly clear, articulated notes, but what if we wanted to overlap those a bit? Try changing your value for duration to 1.5 or even 2 or 3 and listen to the result. If you find the sound is starting to sound distorted as more and more waveforms pile on one another, try turning your amplitude down from 20000.

Let's pause for a second and talk about amplitude in more detail. By now, you know that when we refer to

```
rtsetparams(44100, 2)
load(“WAVETABLE”)

start = 0
duration = 3
amplitude = 10000
note = 48
pan = 0.5
waveform = maketable(“wave”, 1000, “buzz”)

//two octave whole tone scale
for(start = 0; start < 14; start = start + 1){
     WAVETABLE(start, duration, amplitude, cpsmidi(note), pan, waveform)
     note = note + 2
     }
```

Score file 7: Whole tone scale with overlapping notes

Each of the previous two scores contains a slight click, or pop, at the onset of each note, and another at the point where the note stops playing. This is due to the fact that we're literally asking RTcmix to generate a pitch with robust amplitude and one that is without any semblance of shape, or envelope. For example, a piano is unable to gradually fade into a note (unless each string inside the instrument is bowed) due to the sheer mechanics of the hammer striking the strings each time we depress a key on the keyboard. RTcmix is more or less acting in the same way for us right now. Unlike a piano, RTcmix is able to sculpt sounds using various envelope shapes. If each sound has, according to the ADSR envelope, a particular attack, decay, sustain, and release time, we will be able to manipulate our sounds using RTcmix and its maketable( ) command.

We'll start by having each individual note rise to its peak amplitude over a gradual amount of time, rather than start with full amplitude right away, and fade out. Let's do this in the shape of a triangle, so that our sound will fade in

half way through its total duration (since we're using 3, it will fade in over 1.5 seconds) and immediately fade out over the remainder. We can think of this in terms of a graph, with the x-axis representing time and the y-axis designating amplitude, from 0.0 to 1.0.

To draw our envelope using maketable( ), use the following code, which specifies a graph using line point segments:

```
envelope = maketable("line", 1000, 0,0, 0.5,1.0, 1.0,0)
```

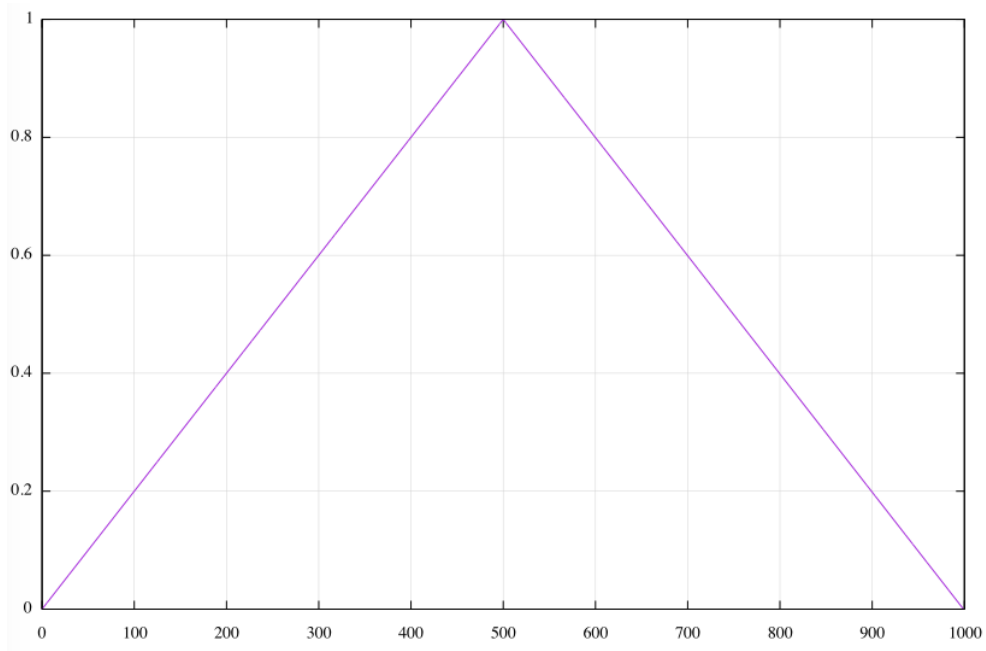

Figure 3: Triangular envelope

You'll note that we're creating a variable called "envelope" with 1000 points. Instead of designating "wave" as we did before, we'll use "line" to give us the line segment graph. What follows are pairs of numbers, in this case 0,0, 0.5,1.0, 1.0,0. I deliberately added spaces between each pair of numbers, which is a stylistic trait that isn't necessary to use, but one that I hope you'll also adopt for line graphs, as it makes understanding the segments that we are using much clearer. In essence, I'm stating that at the beginning of my graph, point 0, use a value of

0. Then, half way through the graph at point 0.5, declare a value of 1.0 (full amplitude). Finally, at the end of the graph, point 1.0, use 0, which is no amplitude, thus no sound.

Something seems fishy with this, however. Although I am using 1000 points in my graph, I'm using points 0, 0.5, and 1.0. Wouldn't it make sense to use points 0, 500, and 1000? Unless you use a special designation for maketable("line") called "nonorm", RTcmix will always normalize your values to something between 0.0 and 1.0. So, even if we did use 500 and 1000, the program will still interpret that as 0.5 and 1000. While it might be confusing at first, I find this pretty useful, especially when using graphs that have a large number of points. Here is our whole tone scale, which uses a graph for sculpting our notes using a standard ADSR envelope.

```
rtsetparams(44100, 2)
load("WAVETABLE")

start = 0
duration = 3
amplitude = 10000
envelope = maketable("line", 1000, 0,0, 0.3,1.0, 0.5,0.5, 0.8,0.5, 1.0,0)
note = 48
pan = 0.5
waveform = maketable("wave", 1000, "sawup")

for(start = 0; start < 14; start = start + 1){
     WAVETABLE(start, duration, amplitude*envelope, cpsmidi(note), pan, waveform)
     note = note + 2
     }
```

Score file 8: Adding an ADSR envelope

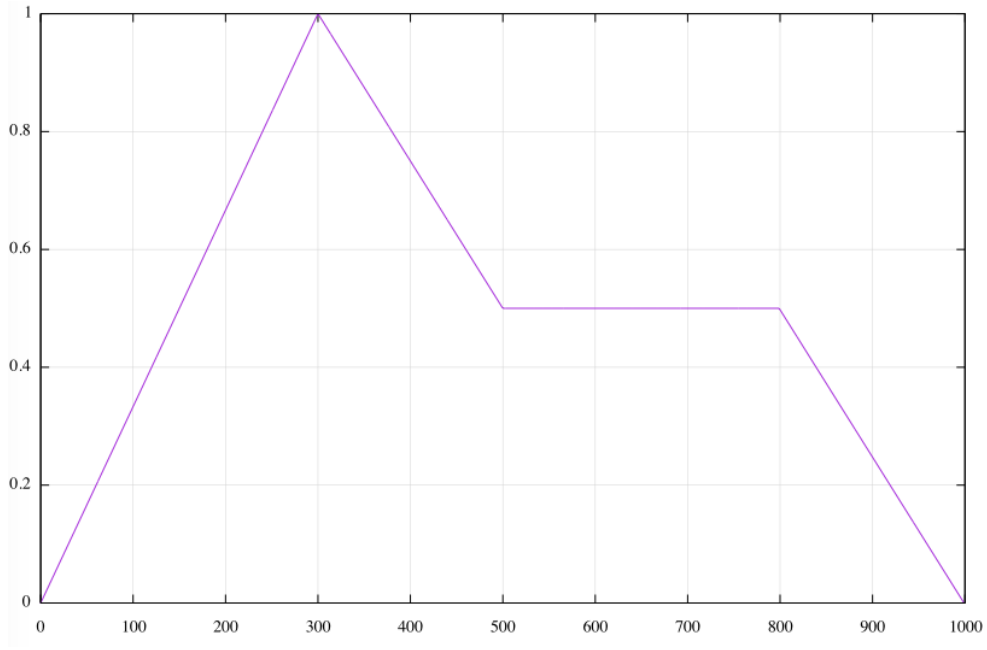

Figure 4: ADSR envelope

In just a few scripts, we've covered quite a bit of ground. Intrinsically, RTcmix — when used on the command line — requires that we set our parameters for sampling rate and the number of channels for audio output. From there, we load our instruments in question and have the chance to declare variables to fill in the respective instruments' p-field values. Most p-field parameters are absolutely necessary, while others are optional. By using the WAVETABLE( ) instrument, we were able to use a variety of waveforms through the maketable("wave") command. Further, when put into a loop structure, we were able to repeat commands, thus only calling on WAVETABLE( ) once in our score file, but also incrementing pitch values - which can be represented in any of a variety of ways - as we moved along. Lastly, we sculpted our individual pitches with the ADSR envelope, utilizing maketable("line").

Here is one final score file that introduces a bit of randomness to our palette and you are encouraged to research the commands that might be unfamiliar to you and understand how they are working in the script.

> RTcmix's website's documentation is a great place to start seeking out this information. You'll note that in this

```
rtsetparams(44100, 2)
load(“WAVETABLE”)

amplitude = 10000
envelope = maketable(“line”, 1000, 0,0, 0.1,1.0, 0.8,1.0, 1.0,0)

increment = 1.0
for(start = 0; start < 50; start += increment){
     duration = irand(4,10)
     note = trand(60,72)
     pan = random()
     WAVETABLE(start, duration, amplitude*envelope, cpsmidi(note), pan)
     increment = irand(0.25,4.0)
     }
```

Score file 9: Chromatic sine wave bath

Sonata II: Random pitch melody

Create a 30 second etude that uses at least two different graph shapes for envelopes, as well as two different ways to represent pitch. Then, randomly select pitches for the melodic material inside of a loop. Don’t feel constrained to only use WAVETABLE( ), but scour the documentation in order to find a new instrument to use in this sonata.

|| Interlude II: Exploring maketable( ) ||

As we've already seen, maketable( ) is very useful in the sculpting and refinement of our sounds and compositions. It has the ability to generate waveforms, graph line segments or curves, import data from text files, create tables from weighted, randomly distributed numbers, and generate window functions, among other useful possibilities. We'll be exploring most, but not all, of the maketable( ) commands in detail here, so the documentation found online will always be your best guide to utilizing any of them in the future.

Filling a graph with line segments is a quick, easy to understand, and helpful way to manipulate data in your scripts. In score file 8, we were manipulating amplitude data, creating a standard ADSR envelope. When we consider that the outgoing amplitude of our MAKETABLE( ) had a peak of 10000, it's easy to understand that if we wanted to have no amplitude, or 0, we would multiply 10000 * 0. Moreover, by subjecting our peak amplitude to the strictures of the ADSR envelope, we were able to generate streams of numbers that altered our volume during playback.

We'll find later on, especially when used in conjunction with Pure Data or Max, that we can further manipulate amplitude in real-time (perhaps even with your iDevice...), but it's important for now to understand that maketable( )'s versatility and easy to understand syntax will greatly enhance your experience with RTcmix and the sounds you are generating.

I can't imagine a world where every detail is visually composed of straight line segments and thankfully the world of maketable( ) isn't limited to "line" either. Let's amend score file 8's envelope shape from the standard ADSR envelope to a triangle shape that has curved line segments in its place.

```
envelope = maketable("curve", 1000, 0,0,5.0, 0.5,1.0,-5.0, 1.0,0)
```

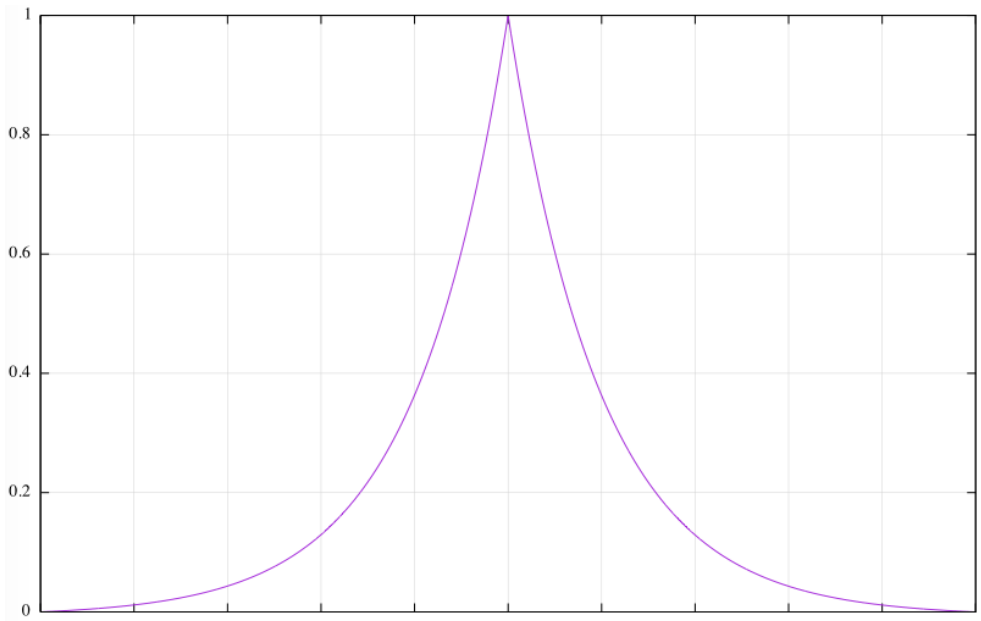

Figure 5: Curvy triangle envelope

Rather than working with pairs of numbers, as we did when using "line", we are instead working with three sets of numbers for each plot, which as a syntactical preference I've spaced accordingly in my list. What we have for each segment in "curve" is a value for x, a value for y, and a curvature value. In essence, I'm asking for the same 0,0, 0.5,1.0, 1.0,0 sequence of points as before, but I'm adding curvature values, in this case 5.0 and -5.0. You can think of it in this way,

```
graph = maketable("curve", 1000, x1,y1,curve1, x2,y2,curve2…, x,y)
```

noting that at the end of our series of numbers, we need only call on a single (x,y) pair with no value for curve, thus the 1.0,0, which denotes our final target value. In our example for the envelope, we used both positive and negative values for the curvature: Negative values will generate convex curves and positive values will create concave curves. The greater the values that you use, the more pronounced your curves will be. Take the envelope example that we generated and try playing around with the values, changing some from positive to negative and back, adding more line segments, using 0 as a curve value (no curve at all, or a straight line), or other transformations that will help you to better understand the possibilities that "curve" will provide for you and the shapes that you can generate with it. Moreover, pause for a moment in order to read through the documentation for "spline" and see how it relates to "curve", and when you're done with that, read how "linebrk" relates to "line".

The graphs that we are generating aren't limited for use with envelopes or transformations in amplitude. Because many of the p-field parameters for many RTcmix instruments can be updated dynamically through tables, it stands to reason that we could use maketable( ) for them as well. In the following score file, we'll explore frequency modulation through the FMINST( ) instrument, which performs FM synthesis using a carrier frequency, modulator frequency, and index. Rather than perform the most basic form of FM, in which we have static values for its requisite components, we'll update them using three different types of maketable( ): "line", "curve", and "expbrk". Moreover, RTcmix's FMINST( ) is unique in that you specify a minimum and maximum index value, while denoting — in p-field 9 — how the index values are being implemented in real time. In the following example, we're sweeping from 1.0 to 10.0 exponentially via the maketable("expbrk") command.

```
rtsetparams(44100, 2)
load("FMINST")

start = 0
duration = 30
amplitude = 10000
carrier = maketable("line", "nonorm", 100, 0,25.0, 0.25,100.0, 0.75,10000.0,
1.0,20.0)
modulator = maketable("curve", "nonorm", 10, 0,15.0,-5.0, 0.75,28.0,5.0, 1.0,0)
index_low = 1.0
index_high = 10.0
pan = 0.5
waveform = maketable("wave", 1000, "sine")
index_envelope = maketable("expbrk", 1000, 0,1.0, 1.0,999.0, 1000)

FMINST(start, duration, amplitude, carrier, modulator, index_low, index_high, pan,
waveform, index_envelope)
```

Score file 10: Dynamic updates for FM

Note that for “expbrk”, we are constructing small exponential line segments that will take on the look of an overall exponential curve. You’ll see that, just like “line”, we’re writing these in pairs, so that for example, I’m setting my first value, or point on the graph’s x-axis, to 0 and the y-axis to 1. This is because all values in “expbrk” need to be > 0. Now, we’ll draw our curve to point 1.0 on the x-axis using 999 tiny exponential line segments to create the curve. Lastly, our target value is 1000, noting that the number of points you specify to be written needs to equal the number of points you delineate for the graph itself.

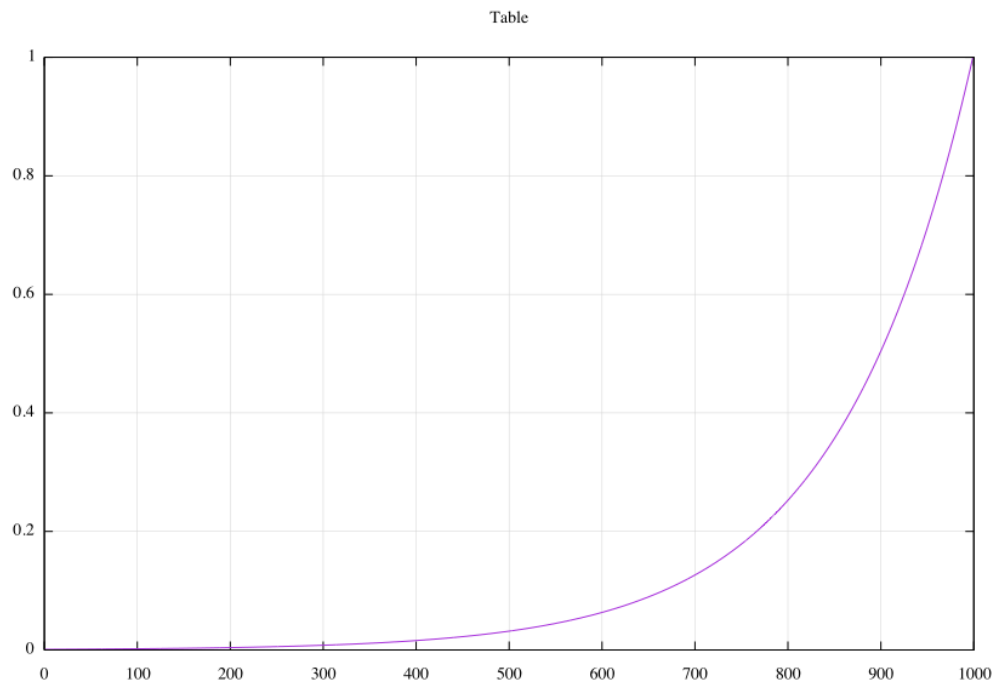

Figure 6: FM index using exponential table

You probably noticed that we called on "nonorm" in two places in our score file. The "nonorm" specifier will ensure that elements in your table won't be normalized to the range of 0.0-1.0 or -1.0 to 1.0, which again is the standard, default setting for all calls to maketable( ). We need this for the carrier and modulator frequencies, less they get normalized to the 0.0-1.0 Hz subsonic range, which could be fun for experimentation.

It is also possible to create tables from the data found within an audio file. You'll need to use maketable("soundfile"), along with the number of points and a path to your file.

```
input_table = maketable("soundfile", 1000, "path/to/file.wav")
```

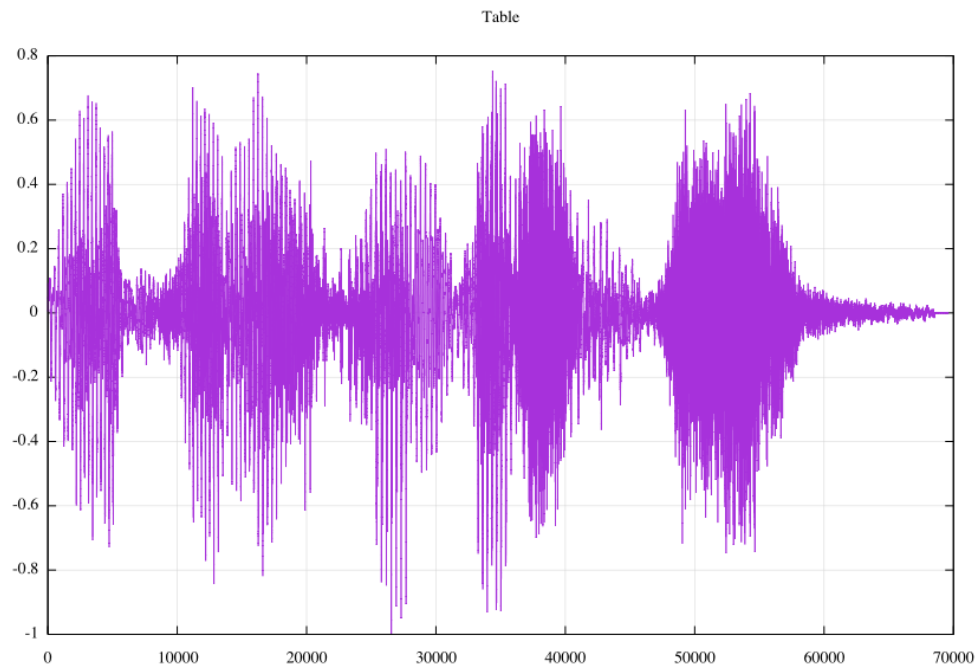

Figure 7: Importing audio file to maketable( )

Thus far, we've kept one important parameter for our sounds quite vanilla. Because our pan values have been set to 0.5, all of our sounds have been reaching us in the middle of the stereo field at all times. RTcmix interprets panning using values of 0.0 (hard right) to 1.0 (hard left) and more often than not pan values can be dynamically updated, so we can use maketable( ) to alter our sounds in time.

It is important to remember that at its base, we are just manipulating raw data using maketable( ), so it is up to you how you'd like to translate that raw data into values that are useful for the instrument at hand. While any of the various types of maketable( ) commands will work for pan values, for example, one very nice feature of RTcmix is the makeLFO( ) command, which oscillates between low and high input values in a way that follows a low frequency oscillator. For example, if we want to pan between 0.0 and 1.0, we would need to call on makeLFO( ), choose a waveform and a corresponding frequency, and our low value (0.0) and high value (1.0). Thus, we'll sweep between our range of values us-

ing a 5 Hz sine wave. This gets particularly fun with a snappy triangle or sawtooth wave.

```
pan = makeLFO("sine", 5.0, 0.0,1.0)
```

In order for your oscillator to truly act as an LFO, specify a frequency that is less than 20 Hz, which is the

One nice addition to makeLFO( ) is the fact that the p-field for frequency can be dynamically updated! So, it's possible to change how fast or how slow the low frequency oscillator is causing our sounds to sweep across the stereo field. You might even try dynamically updating the range of values that makeLFO( ) is sweeping through?

```
pan_frequency = maketable("line", "nonorm", 10, 0,5.0, 0.25,15.0, 0.625,2.5,
1.0,5.0)
pan = makeLFO("tri", pan_frequency, 0.0,1.0)
```

While we teased the notion of randomness using random( ), irand( ), and trand( ) earlier, there may be times when a table of random numbers is more desirable. For example, rather than sticking with the standard set of waveforms available through maketable("wave"), we can can create our own waveforms that are constructed randomly.

Hopefully you were able to successfully uncover the workings of both of these random number generators us-

Because a waveform oscillates between moments of positive energy transfer (compression) and negative energy transfer (rarefaction), we construct waves using both positive and negative numbers between 1.0 and -1.0.

If you'd like to bookmark this page and skip ahead to Interlude V, please feel free to do so. In that interlude,

```
waveform = maketable(“random”, 10, “even”, -1.0,1.0)
```

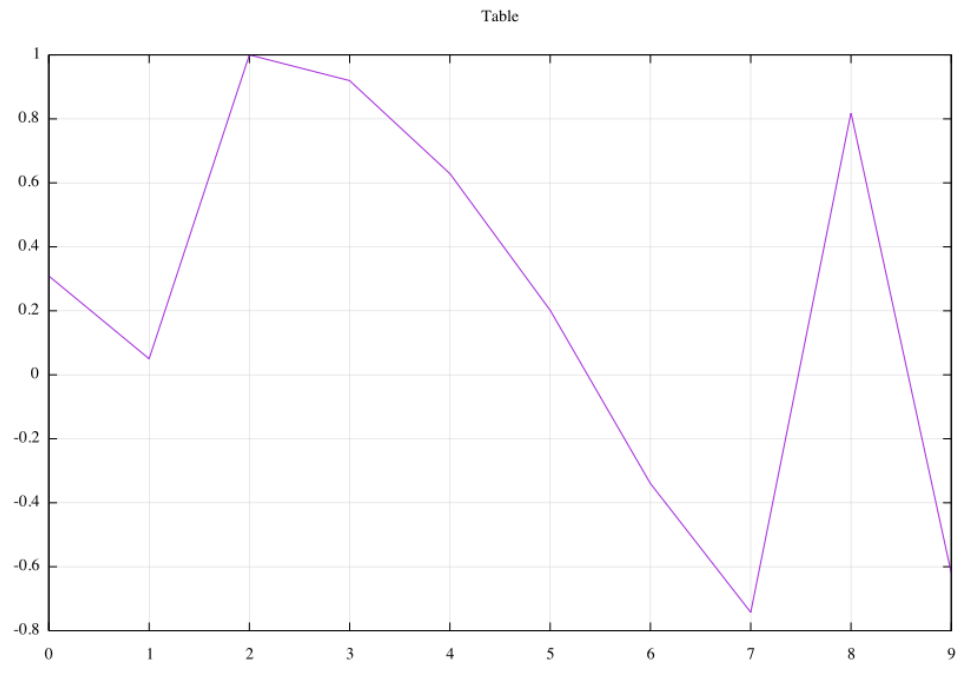

Figure 8: Table of random values

We need to declare the command maketable(“random”) with the following p-fields: The number of points on the graph (10), the type of distribution for those random numbers (even), and both a low value and high value. The distributions are particularly useful, which allow you to choose random numbers that are weighted. For example, if you wanted to choose random numbers between 0 and 11 (for a twelve tone melody, perhaps?)
but you wanted to choose more often from the lower numbers, you’d specify “low”. Choosing “high” will pick randomly, but more often choose the higher numbers.

Note that we’re only using ten points on this graph. The more and more points we use while constructing ran-

There are certain random distributions that have been proven to choose numbers from interesting shapes and curves. You’ll find them by using “gaussian”, “cauchy”, or my favorite, “triangle”. Each of these will favor numbers toward the

middle of your range, but in ways that look like a bell curve or, well, a triangle. If you are excited about these random distributions or probability in general, Mara Helmuth created a particularly fun “prob” option to use with maketable(“random”). In it, you specify minimum and maximum values, as well as a midpoint. Then, a value for tightness will determine how closely the random numbers are chosen near the midpoint value.

```
low_value = 0.0
high_value = 10.0
midpoint = 5.0
/*
The values for tightness work in this way:
0 = randomly choose between only the low value or the high value
1 = evenly choose between any number in the range specified
Using numbers greater than one (up to 100) will begin to “tighten” toward the mid
point.
If 100 is used, almost all numbers will be at or very near your midpoint value
*/
tightness = 75

random_values = maketable(“random”, 100, “prob”, low_value,high_value,midpoint,
tightness)
```

Each of our tables can be further transformed using the makefilter( ) command, which is especially useful when utilizing one table to determine a variety of parameters in your script. For example, we could choose to generate an array of amplitude and pan values between 0.0 and 1.0, later transforming them to generate frequencies falling in a range of our choice.

```
amplitudes = maketable(“line”, 100, 0,0.0, 0.33,0.75, 0.66,1.0, 1.0,0)
pan = amplitudes
frequencies = makefilter(amplitudes, “fitrange”, 220,1530)
```

In this instance, we need to call on makefilter( ), first directing it to the table in question (amplitudes), the type of transformation that we'd like to accomplish ("fitrange"), and finally a range of values to utilize. Even though our original values fall between 0.0 and 1.0, "fitrange" is able to expand those numbers into a new range of your choice. It is also possible to invert your values around a declared midpoint using "invert", or ensure that your values will stay within a specified range using "clip".

"fitrange" will also work with values between -1.0 and 1.0.

In all, RTcmix offers a variety of ways for you to create tables, lists of data, and transformations of those tables or data to enhance your experience, by using maketable( ) and makefilter( ). As a historical note, you might encounter documentation or example scripts that utilize a system of table declarations called makegen( ). This is an antiquated system of creating tables that is no longer being actively updated for future use. It's not out of the realm of possibility that some older RTcmix instruments will still rely upon makegen( ) as the sole way of creating tables, but for the most part, if you'd like to do data transformations, use only maketable( ). In fact, many of the older instruments have recently been revised to use maketable( ), so feel free to try it out. The worst that can happen is an error and maybe you'll be off to the mailing list to either ask about an updated version of the instrument in question or maybe even try updating it yourself? As with anything in this book, the best resource for finding information on any RTcmix instrument, as well as a complete list of maketable( ) and makefilter( ) commands is the documentation found online.

Interlude II: Graph constraints

Compose a 30 second script that uses only one instrument, but every single p-field of that instrument that can be updated using a maketable( ) must be updated. Choose at least two types of maketable( ) and transform one of them using a makefilter( ).

|| Sonata III: Elements of C ||

Computers are able to execute routine tasks in ways that make our lives easier, which includes our compositional work days. In score file 5, we constructed a whole tone scale using separate calls to the WAVETABLE( ) instrument, later streamlining that task into a for( ) loop. Using arrays, nested loops, and conditional tests, we can further enhance our experience with RTcmix and the music we hope to make while utilizing it.

Major and minor scales play a large role in the vocabulary of anyone undertaking the serious study of western art music. For those who might be unfamiliar with the constructs of a major scale, a piano is a great resource. When looking at the keyboard of the piano, you'll notice that it is divided into a series of black and white keys. The black keys are further divided into groups of two and three. Find the grouping of two and play the white note that rests just before the first black note in the grouping of two. You are playing the note C and any white note that falls in the same place along the keyboard will always be the note C. Now, play only the white notes of the piano, ascending, from your original C to the next one and then back down. This is a major scale in the key of C.

There are many, many ways to represent the note C. One is just, well, C, which is a letter designation, useful for reading and writing music. Moreover, there is a special C, called "middle C," which has a frequency of about 261.62 cycles/second, or 261.62 Hz. To further confuse the matter, when using the musi-

cal instrument digital interface (MIDI), we represent that same middle C by the MIDI note 60. Not to be outdone, middle C is also referred to as C4 when using octave designators. To make things even more confusing, RTcmix offers octave point pitch class designation, for which middle C is 8.00. To wit:

> If you were left feeling confused at the piano, unable to find the note C in the first place, a quick online

```
// middle C
frequency = 261.62  //in cycles per second, or Hz
note = 60  //this is the MIDI note number
pitch = C4  //the letter and octave designator
pitch_class = 8.00  //the octave, followed by the pitch. 8.01 = C#, 8.02 = D, etc.
```

In order to construct a C major scale, we have a variety of options in RTcmix. The notes of a C major scale (which you can say aloud to yourself while playing them, ascending, on the piano) are C D E F G A B, followed by the next C, which will put you in the next octave. Since each of those notes can be thought of as an element and all of the elements together can be thought of as a list of elements, we can put them into an array, which we'll do using their MIDI note numbers.

```
c_major_scale = {60, 62, 64, 65, 67, 69, 71, 72}
```

Each element in our list is part of the major scale, which we've called "c_major_scale". It is important to remember that each of our elements in the array (designated as the elements within the curly braces and separated by commas) is also kept at a specific place within the list, or array, itself. Let's pick out just the note E in our list, which is MIDI note number 64.

```
note_e = c_major_scale[2]
```

Take a moment to think about how this is working and don't be afraid if some confusion ensues. We can see that we made a variable called "note_e" because we wanted to grab the note E from our array. Next, we called on the array named "c_major_scale" because that's where our particular note lies. Finally, you'll note the brackets and the number 2: This is the address for our element within the array. This is where confusion might onset.

It looks to me as if E (64) is the third element in our array, which begs the question: Why did we call on 2? Wouldn't that return the second element, D (62)? While we do have eight elements in our array, they are numbered from 0 - 7, rather than 1-8, so you would say that C (60) is the 0th element in the array, D (62) is the 1st element, etc.

```
        [ 0  ,  1 ,    2   ,  3  , 4  ,   5  ,   6   , 7  ]
array = {alpha, beta, charlie, delta, echo, foxtrot, hotel, golf}
alpha = array[0]
echo = array[4]
golf = array[7]   //even though its technically the 8th element.
```

Arrays are particularly useful when put into a loop, as we can cycle through their elements in the same way as playing the ascending notes for C major on the piano.

```
rtsetparams(44100, 2)
load("STRUM2")

pitch_array = {8.00, 8.02, 8.04, 8.05, 8.07, 8.09, 8.11, 9.00}  //C major
// in octave-point pitch class
pitch_length = len(pitch_array)
```

```
//the len( ) command returns the number of elements in the array (8)


//———STRUM2 p-fields
start = 0
duration = 1.0
amplitude = 30000
envelope = maketable("window", 1000, "hanning")
pitch = pitch_array[0]
//can you guess which number we're calling on at this point?
squish = 1.0
decay = 1.25
pan = 0.5

for(start = 0; start < pitch_length; start += duration){
     STRUM2(start, duration, amplitude*envelope, pitch_array[start], squish, decay,
            pan)
     }
```

Score file 11: Ascending major scale

Because our value for "start" will increment by one each time through the loop, it is an incredibly useful way to play through elements in an array, provided that you move from left to right. However, what if we wanted to play the scale both forward and backward, like we might do when warming up on our instrument?

We're using the STRUM2( ) instrument for this next score file as well, which is a synthesized guitar that

```
rtsetparams(44100, 2)
load("STRUM2")

pitch_array = {8.00, 8.02, 8.04, 8.05, 8.07, 8.09, 8.11, 9.00}
pitch_length = len(pitch_array)
```

```
start = 0
//save “pitch” for later
duration = 1.0
amplitude = 30000
squish = 1.0
decay = 1.25

for(iteration = 0; iteration < 1; iteration += 1){

     for(index = 0; index < pitch_length; index += 1){
               pitch = pitch_array[index]
               pan = pickrand(0.3, 0.7)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + duration
                }

       for(index = index - 1; index >= 0; index -= 1){
               pitch = pitch_array[index]
               pan = pickrand(0.3, 0.7)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + duration
               }
}
```

Score file 12: Ascending and descending C major scale

We can always use more than one loop in our score file and in the above example, we're using a nested loop, or a loop within another loop. Here, the variable start isn't being used both in the for( ) loop and as a p-field in the instrument. Instead, we're using something else, because we want to have one meta-loop that will initiate the events for the other loops found inside of it. Our meta-loop gives us the number of times, in total, that we want each of our inner loops to go through. Since I like to think of these as iterations, I used iteration as a

It's common to encounter loops that abbreviate “iteration” for simply “i”. Thus, many programming examples

variable for the meta-loop, which you'll note only happens once.

Within the meta-loop are two more for( ) loops, one to play our scale ascending, and the other, descending. Again, we don't want to use start as our incremental variable in the loop, since we'll be decrementing that value for the descending scale. Instead, we can use a variable called index, which determines the index value of our array, or the particular note being played. With all of this in mind, it becomes clearer how the value for index is being treated, even if we haven't seen the second loop structure, which will start at index value 7 and count down from there.

> Here's another example where understanding the index value in an array is important. We needed to state

Here is a clearer picture of the value for index as we move from top to bottom in this loop.

In the first loop, index will equal: 0, 1, 2, 3, 4, 5, 6, 7

Through the second, it will equal: 7, 6, 5, 4, 3, 2, 1, 0

Now we can see how we're navigating our way through the loop, but what about that value for start times? Since we didn't include it as the incremental value in the loop, we need to work with it somewhere else, which we've done in the last line of each respective loop structure. Since duration equals one, start will equal itself plus one each time. If we play our major scale both ascending and descending, can you figure out the corresponding values for start as we move along?

> Here's a cool hint: Try placing print(start) inside each of the loops and RTcmix will print out that value

We can also transpose our major scale script, in order to include all twelve keys.

```
rtsetparams(44100, 2)
load("STRUM2")

pitch_array = {8.00, 8.02, 8.04, 8.05, 8.07, 8.09, 8.11, 9.00}
pitch_length = len(pitch_array)

start = 0
duration = 0.125  // new duration, akin to sixteenth notes
amplitude = 20000
envelope = maketable("window", 1000, "hanning")
pitch = pitch_array[1]
squish = 1.0
decay = 1.25

transpositions = {0.01, 0.02, 0.03, 0.04, 0.05, 0.06, 0.07, 0.08, 0.09, 0.10, 0.11, 1.00}
num_transpositions = len(transpositions)

for(iteration = 0; iteration < num_transpositions; iteration += 1){

     for(index = 0; index < pitch_length; index += 1){
          pitch = pitch_array[index] + transpositions[iteration]
          pan = pickrand(0.3, 0.7)
          STRUM2(start, duration, amplitude*envelope, pitch, squish, decay, pan)
          start = start + duration  //change durations here
          }

// omit repeating top and bottom notes at the octave
for(index = index - 2; index >= 0; index -= 1){
           pitch = pitch_array[index] + transpositions[iteration]
          pan = pickrand(0.3, 0.7)
          STRUM2(start, duration, amplitude*envelope, pitch, squish, decay, pan)
          start = start + duration
           }
}
```

Score file 13: Transpositions

Data that is placed within any loop in an RTcmix script can also be subject to conditional tests, which are constructs that we find examples of in our everyday language. For example, we might commonly express to someone, "If it's raining outside, then I'll grab my umbrella." In the world of RTcmix and MINC, we can represent this using if( ).

```
if(it_is_raining){
    grab_the_umbrella
    }
```

Conditional tests such as these place the condition inside of parentheses and the outcome of that test within curly braces. There is no limit to the number of outcomes that you might use, so they can be quite powerful.

Even more exciting are logical tests, which enhance how we utilize our conditionals. For example, we might pose this question: "Choose a random integer between 0 and 10. If the random number is greater than or equal to 5, turn up my amplitude by 0.25, or else turn it down by 0.25."

```
srand()

x = trand(0,10)
printf("the variable x equals: %f \n", x)

amplitude = 0.75

if(x >= 5){
     amplitude = amplitude + 0.25
     }

else{
     amplitude = amplitude - 0.25
     }

printf("amplitude now equals: %f \n", amplitude)
```

Important note about the use of srand( ) in the opening line of this script. Each equation that returns a

printf( ) is more of a C-style printing. You'll see the results in your Terminal window.

Score file 14: Conditional example

Even though this seems like a rather vanilla conditional example, it has the elementary constructs of a random walk. Random walks are processes that either increment or decrement according to a given probability.

Let's use pitch as a variable we can subject to a random walk procedure. We'll set up the random walk so that the pitch of our instrument will either increase by a half step or decrease by a half step equally. Think of it this way: A two sided coin has the equal probability of landing on either heads or tails, so if we flip a coin 100 times and take note of all the heads and all the tails, we can reasonably expect to have an outcome close to 50 heads and 50 tails. Similarly, we'll do a conditional test that reasonably ensures we increment 50% of the time and decrement 50% of the time.

```
rtsetparams(44100, 2)
load("WAVETABLE")
srand()

start = 0
duration = 0.25
amplitude = 18000
envelope = maketable("curve", 1000, 0,0,1, 0.1,1.0,-10, 0.33,0.5,-1, 0.8,0.25,-10, 1.0,0)
note = 60
pan = random()
waveform = maketable("wave", 1000, “tri”)

increment = 0.0625
for(start = 0; start < 100; start += increment){
     WAVETABLE(start, duration, amplitude*envelope, cpsmidi(note), pan = random(),
     waveform)

     x = random()
     if(x < 0.5){
```

```
     note += 1  //50% of the time, increase by a half step
          }

 if(x > 0.5){
          note -= 1  //50% of the time, decrease by a half step
          }
}
```

Score file 15: Random walk flight of the bumblebee

Using the conditions put forth in this score file, it's possible to change the probability of the outcomes. Because "x" is returning numbers from 0.0 to 1.0, it's useful to think of those as percentages, thus the 0.5 condition. However, if you'd rather have the melody that slightly favors an ascent, change that number to 0.52 or 0.55, remembering to correspondingly change the values for decrementing to 0.48 and 0.45, respectively. What results in this score is something of a highly wavering, chromatic flight of the bumblebee.

Conditions aren't limited to phrases such as "if x is less than" or "if x is greater than or equal to". Remember our first example when we used "If it's raining, then..." as a conditional test? Well, what if we wanted to subject our outcome to more than one condition, such as "If it's sunny or if it's raining, then..." For this, we'll need to add some more symbols to our tool belt.

```
if(it_is_sunny || it_is_raining){
    grab_the_umbrella
    }
```

Note how we had to use two parallel lines (| |) to complete our statement. This is another example of an operator that we can utilize for conditional tests in C.

```
<   less than
>   greater than
<=  less than or equal to
>=  greater than or equal to
==  is equal to

!=  is not equal to
&&  and, such as if(x > 10 && x <=20)
||  or, such as if(x > 10 || x <= 20)
```

When defining variables, use =, such as alpha = 100. In a conditional test, use ==, such as asking if(alpha ==

We can also test to see if a number is even or odd, by checking to see if there is a remainder when a given number is divided by two. This works well for the following example (a personal favorite), which outlines the "wondrous number" algorithm laid out by Douglas Hofstadter in his book *Gödel, Escher, Bach: An Eternal Golden Braid*.

```
rtsetparams(44100, 2)
load("MMODALBAR")
srand()

/*
3n + 1 algorithm = a simple procedure for generating data
"The Wondrous Number algorithm"

Step 1: Take a random number.
Step 2: If the number is ODD, triple it and add one.
Step 3: If the number is EVEN, divide by two.

A wondrous number will divide itself all the way down to 1
and repeat over and overand over.
*/

n = trand(100,127)
print(n)

for(i = 0; i < 5; i += 0.125){
```

You'll want to run this script multiple times to get the full effect of the subtle differences, especially the differ-

My piece *A Wondrous Number* relies on this algorithm.

```
if(n%2 == 0){  //if n / 2 DOESN'T leave a remainder (is even...)
     n = trunc((n / 2)) //trunc() takes off any decimals
     }

else if(n%2 == 1){  //if n / 2 DOES leave a remainder (is odd...)
     n = trunc(((n * 3) + 1))
     }

print(n)

MMODALBAR(start = i, duration = 0.125, amplitude = 20000,
frequency = cpsmidi(n), hardness = 1.0, position = 0.5, 0, pan = 0.5)
}
```

Score file 16: Wondrous Number 3n + 1

Understanding these principles of the MINC parser, which are so closely tied into the C programming language, will greatly enhance your experience with RTcmix. In fact, the concepts covered in this sonata form the essence of the program itself and separate it from so many proprietary software programs available to us today.

Most computer music audio workstation programs, be it Ableton Live, Reason, Pro Tools, Reaper, or Logic, contain plugin effects that modify your sounds. However, RTcmix affords us the ability to algorithmically design how we use those sounds and effects in time using a programming language, rather than the admittedly gorgeous (you have to pay for something, right?) user interface of the aforementioned programs. When we encounter those effects and sounds in syntax, we're more apt to understand and research the parameters that define them, all while sculpting and refining those sounds using unique processes. More than one student has remarked, after learning a bit of RTcmix, how much their experience with proprietary DAWs has been enhanced after realizing exactly how those GUI

plugin effects are actually functioning. Brad Garton said it best and I couldn't agree more: When it comes to composing electronic and computer music, "...I generally think the way RTcmix does."

Here is one last example score file that relies heavily on conditional tests. In it, I'm creating a drum machine from individual, mono drum set samples (bass drum, snare drum, hi hat, etc.) that are arranged in an array from low sounds to high sounds. From there, I'm using a variety of conditional tests to first choose from the outermost (extreme) elements of the array (so this should rely more heavily on bass drum and hihat sounds) and gradually shift toward choosing from the middle of the array (tom sounds and snare), eventually randomly selecting from any part of the array. We haven't looked at the JFUNCS( ) instrument (really a library of functions) yet, but I'm loading it in order to utilize map( ), which takes incoming numbers in a range of 0 to 1 and expands them to another range, specified by the user. In this case, I'm expanding 0 to 1 to the range of 0 to 5 to randomly select from the drum sounds in my array. Moreover, I need to use trunc( ) to take off the decimal point that will result from map( ), since array locations can't intrinsically be floating point numbers.

If I wanted to truly round these numbers, I'd need to do something like

```
index = (trunc(map(outcome, 0,1,
```

```
rtsetparams(44100, 1)
load("DELAY")
load("JFUNCS")
srand()

bd = "/path/to/BD01.wav"
sd = "/path/to/SD01.wav"
tomlow = "/path/to/TOM05L.wav"
tomhigh = "/path/to/TOM09H.wav"
ride = "/path/to/RIDE1.wav"
hihat = "/path/to/CLHAT1.wav"

drums = {bd, tomlow, tomhigh, sd, ride, hihat}
numdrums = len(drums)
```

```
//drums arranged from low sounds to high sounds
//begin by tending toward extremes, then middle, then all

index = 0
increment = 0

for(start = 0; start < 50; start += increment) {
     x = random()
     y = random()

          if(start < 10){
               if(x > 0.75){
                    outcome = x
                    index = trunc(map(outcome, 0,1, 0,numdrums))
                    }
               if(y < 0.25){
                    outcome = y
                    index = trunc(map(outcome, 0,1, 0,numdrums))
                    }
               }
               else{
                    index = pickrand(0,1, 4,5)
                    }

     if(start > 10 && start < 20){
          outcome = ((x + y) / 2)
          index = trunc(map(outcome, 0,1, 0,numdrums))
          }

     if(start < 20){
          index = trand(numdrums)
          }

file = drums[index]
rtinput(file)
DELAY(start, instart = 0, duration = DUR(), amplitude = 1.0, delaytime = DUR())
increment = DUR()
}
```

Score file 17: Random drum machine

Sonata III: Conditionals

Search the documentation for an instrument that hasn't been used or discussed thus far and find as out as much as you can about it. Then, create a 1' score file that comments on each of the p-fields that constitute the instrument, generates values from an array, and uses at least three different conditional tests to alter parameters.

|| Interlude III: RTcmix and Python ||

In the RTcmix world, MINC is sometimes referred to as a parser, front end, or language. Any of these terms refer to the fact that MINC code (which again, borrows from C) is the syntax used to create our score files. However, MINC isn't the only language that we can utilize in the course of our work with RTcmix: Both the Perl and Python languages have been ported to RTcmix, so that we can compose our scripts using their constructs, rather than those of MINC. For some, this interlude is likely going to be completely useless, as they'll be more than happy to continue working in MINC. For others who might already be familiar with Python, this will be a fun interlude to look through. Because I enjoy using the Python programming language, this interlude will focus on some of the basics of the language and how it can be used in conjunction with RTcmix.

Are there advantages to using Python with RTcmix? Yes and no. Unlike MINC, Python is a so-called object-oriented programming language, which allows the user to define classes of "objects" that can be used throughout the program. Moreover, when using Python, we can import all of its distinct libraries to further refine our scripts.

> The purpose of this interlude isn't to discuss Python in full, so please don't rely on it as a primer to the world of

However as of this writing, creating RTcmix scripts using a feature-rich language like Python isn't supported when using the [rtcmix~] object in Pure Data or Max.

As noted in the installation process, it is possible to configure your build of RTcmix using Python, among other ancillary features. We'll can save our score files with the suffix .sco, or you can save them with .py, which denotes a Python file. Either way, we can still play them in our Terminal using:

```
pycmix < /path/to/your/file.py
```

Our first step in a script that uses Python isn't to call on rtsetparams( ), but instead to write

```
from rtcmix import *
```

which will add all the elements of the "rtcmix" namespace to our script. Think of this as an RTcmix library working within Python, which ensures that Python knows what to do with RTcmix-intrinsic tasks like WAVETABLE( ) and PANECHO( ) and bus_config( ), which obviously aren't native to Python itself. From there, we can call on rtsetparams( ) like we did earlier, as well as import( ) for loading instruments. Let's rewrite score file 9 in Python and compare its new syntax with MINC.

```
from rtcmix import *
# note that using "#" is the correct way to comment in Python
rtsetparams(44100, 2)
load("WAVETABLE")

amplitude = 10000
envelope = maketable("line", 1000, 0,0, 0.1,1.0, 0.8,1.0, 1.0,0)

increment = 1
```

```
for start in range (0, 50, increment):
    duration = trand(4,10)
    note = trand(60,72)
    pan = irand(0.0,1.0)
    WAVETABLE(start, duration, amplitude*envelope, cpsmidi(note), pan)
```

Score file 18: Score file 9 in Python

Because MINC intrinsically does away with the semicolons found at the end of each line of code in the C programming language and Python doesn't have any semicolons at all, both scores look strikingly alike. I find Python to be an incredibly elegant way of writing code. In fact, the for loop structure omits the use of curly braces and instead relies on indentation to delineate processes that need to be executed in the loop itself. However, attempting to increment by a floating point number will cause some headaches, as for loops in Python are solely incremented by integer numbers.

Here's a bit of code that will solve the issue for us.

```
def custom_range(start, stop, step_size):
    x = start
    while x < stop:
        yield x
        x+= step_size
```

This is one point where Python becomes powerful and customizable, as we can create our own custom loop

The three constituent components of our object are a start value, stop value, and step size for incrementing. Let's say that we start at 0, have a value of 10 for stop, and increment by 0.5 each time through the loop.

Within our definition, we create an arbitrary variable, "x", that will equal start, since we need to declare start times for our loop structures. Then, as long

as “x” is less than “stop”, yield “x” and increment it by the “step_size”. My hope is that the previous two sentences sound stunningly similar to the process for utilizing for( ) loops in MINC because they should. Note again the use of indentation and how it functions within “custom_range”.

We can also use a so-called while loop, which essentially repeats a set of directions while a given variable - in this case a number - is less than a conditional that we specify. This is going to be much easier to use when incrementing by a floating point number.

```
while start < 10:
    x += 1
    start = random()
```

Here is a script that uses a while loop and a list of elements.

```
from rtcmix import *

rtsetparams(44100, 2)
load("WAVETABLE")

# here's one example of a Python library available for our use
import random
# in order to use an feature of the random library, use random.nameoffeature
# below, we'll use random.random, which works in the same way as MINCs random()

srand()

#----- WAVETABLE p-fields
start = 0
duration = 0.125
amplitude = 20000

# generate a list of MIDI notes, similar to filling an array

my_list = [] # declare the list of elements
```

```
num_elements = 100

for i in range(num_elements + 1):
    my_list.append(random.randrange(50,101,1))
# can you see how the line above works?

print my_list

increment = duration
index = 0

while start < (num_elements / 8):
    WAVETABLE(start, duration, amplitude, cpsmidi(my_list[index]),random.random())
    index += 1
    start += increment
```

Score file 19: Python script with a list and a while loop

With MINC, we work with arrays of elements and in Python we work with lists. In score file 18, we are using a loop to fill a list of elements, my_list, with random integers between 50 and 101, which are in turn being translated to MIDI note numbers to use as a random melody with WAVETABLE( ). Using the append feature allows us to add a new random element to our list each time we cycle through the loop. Moreover, because our duration values are at 0.125 and our incremental value for the loop is equal to that duration, we are hearing our melody in a sixteenth note pattern.

Regarding tempo: If we consider that an incremental value of 1 will intrinsically increment every second, it

Conditionals in Python are also similar in MINC, with the syntactical caveat that again we focus on indentation instead of curly braces. The next example score file doesn't make sound, but instead creates four note chords for use in a piano or pitched percussion part.

```
from rtcmix import *

import random
from random import randrange
random.seed()

lowest_note = 45   # In MIDI
highest_note = 96
mode = highest_note - lowest_note
# Think back to mean, median, and mode. What is mode again?

# Have each voice (soprano, alto, tenor, bass) start on mode value
s = a = t = b = mode

# In the loop, we'll watch each voice move inward and outward

for i in range(0,10):
    x = random.randrange(10) #random number to use for conditional tests
    if x < 5:
        s += 1
    else:
        s -= 1

    if x < 2 or x > 8:
        a += 1
    else:
        a -= 1

    if x < 7:
        b -= 1
    else:
        b += 1

    if s - a > 12 or t - b > 12:
# this is quite slick, writing “or" instead of ||
        s = s - 12
        t = t - 12
```

```
gamut_in_range = [b, t, a, s]
print "gamut in spread range: %s" % gamut_in_range
```

Score file 20: Making chords from conditionals

Essentially, each respective voice is being subjected to its own random walk, going up or down by half steps according to the returned value for x. Moreover, the final if statement ensures that the soprano voice will always be within an octave of the alto voice and that the tenor will always be an within an octave above the bass voice. This ensures that not only do we adhere to the conventions of four-part chorale writing (a favorite of mine), but that if we notate them, future pianists or vibraphone players who play our gorgeous random chord collection won't have to spread their fingers or mallets too far to reach the notes.

One very important advantage of Python over MINC is the ability to create callable objects. They are routine tasks that you can define and call on at any point in your score file after you've created them. For example, let's say that we want to create a function that returns a number squared, such that the number x will equal x times x.

```
def square(x):
    return (x * x)
```

At any point in our score file, we can now call upon our new function, square, by writing it with a corresponding element to evaluate in the same way that we use commands like abs( ), round( ), trunc( ), or wrap( ) in MINC.

```
element_squared = square(4)
```

We could create another function to evaluate the square root of a given element, but this would also be a pretty routine task and one that we'd expect to have available to us at any time. Thankfully, a host of exciting features await those who import Python's math library, just as we imported elements from its random library earlier.

Python is incredibly adept at treating lists, which is a powerful tool when we consider pitch transformations later on in the book. Let's fill a list with six random elements that we will use as pitches.

```
spray_table = 1
spray_size = 6
seed = 12

aggregate = []
spray_init(spray_table, spray_size, seed)
#spray_init is an RTcmix-intrinsic command

for i in range(0,spray_size):
   aggregate.append(get_spray(spray_table))

print(aggregate)
```

The spray_init( ) command comes from RTcmix and returns unrepeated random numbers from 0 up to the range specified. In this case, we're getting random numbers from 0 to 5 that we're using to fill our aggregate list. We can transpose them into a meaningful range, such as MIDI, by creating our own transposing object.

```
def list_to_midi(pitches, range):
    return [x + range for x in pitches]

series = list_to_midi(aggregate, 60)
```

```
print(series)
```

Note that we are first declaring two arguments. The first is the set of pitches that we'd like to transform — in this case, our aggregate list — and a range to transpose them into, such as 60 for the range starting above middle C. From there, we're returning each individual x value plus 60 when we finally utilize our object to create a new list.

For the majority our work, we'll be using MINC. However, for those excited about and interested in the world of Python itself and how it can be used in conjunction with RTcmix, I highly encourage you to write the remaining examples in both languages. To end this interlude, we'll explore a complex Python example and it will be your job to research each of the lines in question and how they are working.

```
from rtcmix import *
import random

rtsetparams(44100, 2)
load("MMODALBAR")

#---------- MMODALBAR
start = 0
duration = 1
amplitude = 30000

#---pitch stuff
spray_table = 1
spray_size = 12
seed = 12

aggregate = []
spray_init(spray_table, spray_size, seed)
for i in range(0,spray_size):
   aggregate.append(get_spray(spray_table))
```

This example will come up again when we explore twelve-tone transformations in Interlude VI, but it's pre-

```
def retrograde(pitches):
     return pitches[::-1] # Search “list slicing in Python” for an explanation

def invert(pitches):
     return [((x - 12) * -1) for x in pitches]

def retrograde_invert(pitches):
     return invert(retrograde(pitches))

def series_to_midi(pitches, range):
     return [x + range for x in pitches]

p_0 = series_to_midi(aggregate, 32)
r_0 = series_to_midi(retrograde(aggregate),48)
i_0 = series_to_midi(invert(aggregate), 60)
ri_0 = series_to_midi(retrograde_invert(aggregate), 72)

hardness = 1.0
position = 1.0
instrument = 4
pan = 0.5
```

```
# series of chords
for start in range(0,spray_size):
   MMODALBAR(start, duration, amplitude, cpsmidi(p_0[start]), hardness, position, instrument, pan)
    MMODALBAR(start, duration, amplitude, cpsmidi(r_0[start]), hardness, position, instrument, pan)
    MMODALBAR(start, duration, amplitude, cpsmidi(i_0[start]), hardness, position, instrument, pan)
    MMODALBAR(start, duration, amplitude, cpsmidi(ri_0[start]), hardness,position, instrument, pan)
```

Score file 21: Dodecaphonic transformations in Python

I find that in our era of unparalleled access to information, it is incredibly useful to seek out answers using known resources. From experience, I can say that learning the basics of Python by rewriting example scripts from others and using

comments to help me better understand the functions of the program was a very enriching experience.

I'm also a proponent of immersive learning styles and building skills incrementally. If even twenty minutes per day are spent rewriting known RTcmix files into Python and taking the time to search concepts that you either haven't encountered before or would like to know more about, you'll soon find a sense of facility with your work and a more enlightening experience using RTcmix while composing your work.

Interlude III: Python and RTcmix

Using score file 20, use the chords that you generate as pitch material for a short piece, which uses an instrument that you searched in the online documentation and haven't yet used in your work.

|| Sonata IV: Adding various effects to your scripts||

Much like a chain of guitar pedals, RTcmix is capable of sending your processed sounds into any of a number of effects to further enhance and refine your musical ideas. Not only does RTcmix contain the standard set of DSP effects — such as delay, various types of filters, EQ, flange, or chorus, and others — but it can dynamically update those effects in real-time, which again yields endless possibilities during the composition process.

We'll begin by sampling a standard audio file and passing it through an effect via the bus_config( ) command.

```
rtsetparams(44100, 2)
load("STEREO")  //stereo sampling, must be .aiff or .wav!

rtinput("/path/to/your/file.aiff")   //remember to point to a file on your computer

start = 0
instart = 0   //input start time
duration = DUR()
amplitude = 1.0  //relative, not absolute
pan = 0.5
STEREO(start, instart, duration, amplitude, pan)
```

Score file 22: Simple stereo playback of a sound file

With STEREO( ), there are two p-field values for start times. The first is the overall start time, which we are familiar with, and designates when to onset the STEREO( ) instrument itself. Again, we want the instrument to start right way, so we choose a start time of 0. Now, STEREO( ) also includes an input start time, or at what point within the audio file to begin playback. For example, if we are working with a 30" sound file and set an input start time of 15, STEREO( ) will play the sound file starting at 15", and then play through to the end. The DUR( ) command returns the overall duration of the audio file, so if we are working with a 30" clip, DUR( ) would return a value of 30. P-field values for amplitude and panning are dynamic, so those can be updated with a table for more exploration.

If we did want to use an input start time of 15, it's good practice to use

As a first example, we'll pass the audio from STEREO( ) to a REV( ) instrument using the bus_config( ) command to add a little bit of reverb to our sound.

```
rtsetparams(44100, 2)
load("STEREO")
load("REV")  //reverb instrument
rtinput("/path/to/your/file.aiff")   //remember to point to a file on your computer

//send audio through an auxiliary line
bus_config("STEREO", "aux 0-1 out")
start = 0
instart = 0   //input start time
duration = DUR()
amplitude = 1.0  //relative, not absolute
pan = 0.5
STEREO(start, instart, duration, amplitude, pan)

//receive audio from aux line 0-1 and send out
bus_config("REV", "aux 0-1 in", "out 0-1")
type = 1     // 1 is Perry Cook's, 2 is John Chowning's, 3 is Michael McNabb's
rvbtime = 2.5
```

RTcmix also includes a Schroeder reverb instrument, REVERBIT( ) and two other reverb instruments worth

```
rvbpct = 0.5
inchan = 0
REV(start, instart, duration + rvbtime, amplitude, type, rvbtime, rvbpct, inchan)
```

Score file 23: Adding reverb with bus_ config( )

Let's have a bit of fun with our audio file before passing it to a delay instrument. We'll first put it through a loop that will pick random start times within the file and play them for 0.25 seconds, which corresponds with our increment value. Then using DEL1( ) — just one of RTcmix's delay instruments — we'll be able to very succinctly take a copy of the sample in question and play it again at a (you guessed it) random amount of time.

```
rtsetparams(44100, 2)
load("STEREO")
load("DEL1")  //delay instrument
rtinput("/path/to/your/file.aiff")

bus_config("STEREO", "in 0-1", "aux 0-1 out")

increment = 0.25
for(start = 0; start < 100; start += increment){
     instart = irand(0,DUR())
     duration = increment
     amplitude = irand(0,1.0)
     pan = 0.5
     STEREO(start, instart, duration, amplitude, pan)
}

bus_config("DEL1", "aux 0-1 in", "out 0-1")
start = 0
instart = 0
delay_duration = 30
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, 0.1,1.0, 0.9,1.0, 1.0,0)
```

```
delay_time = maketable("random", 10.0, "triangle", 1.0, 8.0)
r_ch_amplitude = 1.0 // right channel relative to left channel
DEL1(start, instart, delay_duration, amplitude*envelope, delay_time,
r_ch_amplitude)
```

Score file 24: Delay instrument

The DEL1( ) instrument processes a delayed copy of the original sound file (or input audio source) at a specified delay time in p-field 6, where we've opted to randomize that process using a table of random numbers between 1.0 and 8.0 (seconds). Then, we process the delayed copy in the right channel relative to the original signal in the left channel. The result is a 30 second excerpt of randomly-derived delayed copies of your sound source. There are other delay instruments in RTcmix, including DELAY( ), which processes delayed copies of your signals with a specified feedback parameter. This will afford you to ability create ping-pong style delay lines (think of the phenomenon of the sound of a ping pong ball falling toward a table, having enough energy to spring back up, then down, then up, then down, ever so slightly less and less) and other types of delays according to your own algorithmic procedures.

Let's shift focus on to some of the so-called short-term delay effects, such as echo, doubling, chorus, and flange, all of which are available to us in the RTcmix library of instruments.

Each of these effects involve some type of delayed copy of an input source and differ in the amount of overlap or time processed. Let's first look at PANECHO( ), RTcmix's echo instrument, which takes input audio and not only provides echo, but pans it back and forth across the stereo field in the aforementioned "ping pong" style.

```
rtsetparams(44100, 2)
load("PANECHO")   // an echo instrument that includes panning

rtinput(“/path/to/file.aiff“)
bus_config("PANECHO", "in 0-1", "out 0-1")

//------------------ping-pong delay
start = 0
instart = 0
duration = 60
amplitude = 1.0
envelope = maketable("window", 1000, "hanning")
channel0delay = 0.25  // in sec
channel1delay = 0.50
delay_feedback = 0.8     // ALWAYS less than 1.0 or a crazy feedback loop ensues!
ringdowndur = 1.0

PANECHO(start, instart, duration, amplitude*envelope, channel0delay, channel1delay,
delay_feedback, ringdowndur)
```

Score file 25: Echo with panning via PANECHO( )

Echo is itself a process that delays the signal by about 40 milliseconds or longer and one super important concept to keep in mind with PANECHO( ) is the p-field for feedback, which can get out of hand quickly if it's accidentally set above 1.0. You'll note that the echo plays back in the left channel at a delay time of 0.25 seconds and in the right at 0.5 seconds, which — along with the feedback time — can be updated using dynamic tables.

If a signal is played back with an exact copy of itself that is delayed anywhere from fifteen to 40 thousandths of a second, the overall effect will be a doubling of the original signal. You can verify this phenomena by first playing a file back using STEREO( ) and seeing its resulting maximum amplitude in the Termi-

nal window, then doubling the signal with another STEREO( ) call that is delayed by, say, 0.004 seconds and seeing its resulting maximum amplitude.

```
rtsetparams(44100, 2)
load("STEREO")
rtinput(“/path/to/file.wav“)

start = 0
instart1 = 0
instart2 = 0.004
duration = DUR()
amplitude = 1.0
pan = 0.5
STEREO(start, instart1, duration, amplitude, pan)
STEREO(start, instart2, duration, amplitude, pan)
```

Score file 26: Doubling a signal using STEREO( ) and requisite delay

While doubled signals are of course great to use, RTcmix also includes a wonderful instrument called HOLO( ), which produces a “Carver Sonic Hologram” generator that widens the stereo field. What we’re talking about here with doubling and HOLO( ) is really ensuring that — much like normalization— we’re using the most robust, cleanest, clearest audio source file in our compositions, which can in turn be processed later. Let’s first use HOLO( ) to widen our original file and then send it to the FLANGE( ) instrument.

```
rtsetparams(44100, 2)
load("HOLO")  //phase cancellation
load("FLANGE")

rtinput(“/path/to/file.wav“)
```

```
bus_config("HOLO", "in 0-1", "aux 0-1 out")
start = 0
instart = 0
duration = DUR()
signal_amplitude = 0.5  //amplitude of original audio
processed_amplitude = 0.75 //amplitude of "hologram" of signal
HOLO(start, instart, duration, signal_amplitude, processed_amplitude)

bus_config("FLANGE", "aux 0-1 in", "out 0-1")
amplitude = 1.0
resonance = irand(0,1)  //percentage, 0-1
maxdelay = 1 / cpspch(9.02) //usually determined as 1 / cpspch(octave.pitch)
moddepth = 25 //percentage, 0-100
modrate = makeLFO("sine", 10.25, 1.0,5.0) //in Hz, fun to use an LFO for this
wetdrymix = 0.8 //percentage, 0-1
FLANGE(start, instart, duration, amplitude, resonance, maxdelay, moddepth,
modrate,wetdrymix)
```

Score file 27: Stereo hologram of sound with FLANGE( )

The above score file works really well when processing sounds of speech or other audio sources that have a particularly noisy spectrum, as the flanger's periodic shifting causes cancellation and reinforcement of certain frequencies of your source audio.

There are many, many more instruments and effects to explore in conjunction with bus_config( ). While this Sonata explored short term delay effects in detail, the following interlude will focus specifically on subtractive synthesis and digital filters, all while introducing more complex types of chained connections for further sculpting and refinement of your sounds.

SONATA IV: A brief RTcmix etude

Using at least three delay effects covered in this Sonata, as well as at least two more RTcmix effects that you discover on your own via the online documentation, create a two to three minute etude using chained effects via bus_config( ). See how many "aux in" and "aux out" connections you can make and supplement your etude with a brief paragraph or two detailing your compositional process as you wrote your work.

|| Interlude IV: A detailed look at bus_config( ) and connections ||

In order to begin understanding the connection of instruments in RTcmix in detail, we'll need to begin by first exploring one of the oldest CMIX instruments that is still in use with the program, and little by little begin building complex chains of connections using some of the more recent additions to the suite of RTcmix tools.

As alluded to in the Prelude, Paul Lansky's work with CMIX more or less boiled down to creating a compositional tool that would afford the user the ability to mix together various sound files using the C programming language. This early instrument still exists in RTcmix and you'll be able to use it without having to first load it using load( ) as we've become accustomed to thus far. That is because MIX( ) remains a critical piece of the RTcmix, and as the online documentation states, MIX( ) is just "always there. Always."

We've already seen MIX( ) in action when we used STEREO( ) earlier. This is because while MIX( ) has existed from the beginning, STEREO( ) has more or less taken over as the main tool for audio file sampling in RTcmix. That isn't to say, however, that we can't still use MIX( ), but in order to do so, we need to get used to its channel matrix. In essence, using either of these two instruments gets to the heart of connections in RTcmix, namely, taking an audio file in (by loading it) and sending it out to your sound card.

```
rtsetparams(44100, 2)
rtinput("/path/to/your/file")

duration = DUR()
amplitude1 = makeLFO("sine", 0.3, 0.25,1)
MIX(start = 0, instart = 0, duration, amplitude1, 0, 0)

amplitude2 = makeLFO("sine", 0.4, 1,0.25)
MIX(start = 1, instart = 0, duration, amplitude2, 1, 1)
```

Score file 28: MIX( ) with amplitudes in and out of phase

The above score file takes one input file and mixes it between the left and right channels of the stereo field, delaying one copy of the sound file by one second, the result being something of a canon or *caccia*. The last two p-fields for MIX( ) can be confusing for multichannel diffusion, but for stereo output is pretty straightforward: They delineate the input channel and output channel. So, in the first call to MIX( ), I'm stating that I'd like to have input channel 0 (from my audio file) sent to output channel 0. That's pretty simple to understand, and you can deduce what 1,1 means in the last two p-fields for the second MIX( ) call. One particularly useful strategy for MIX( ) is to use -1 as an output "destination" in order to mute that channel.

Most readers are probably going to be using RTcmix in stereo, however, there are a number of instruments

Each time you send a sound out of an auxiliary line to another instrument, you are losing its original constructs, which you might want to use at another point in your score file. For example, think about a call to WAVETABLE( ) that is being sent via "aux 0-1 out" to FLANGE( ), which is then sent out to your sound card. Your original waveform or whatever cool processes that you were doing with WAVETABLE( ) alone will forever be subject to flanging, and we'll

cease to hear the original waveforms themselves. There may be times when you might want to hear both the original sounds and their flanged results at the same time, but we can't send a sound out to two places simultaneously using bus_config( ).

```
bus_config("WAVETABLE", “aux 0-1 out”, “out 0-1”)  //ERROR!
```

To mitigate this dilemma, we can use SPLITTER( ), which affords us the ability to circumvent the problem of the “one in and one out” methodology of bus_config( ). In essence, SPLITTER( ) takes inputs and sends them out to multiple outputs, with the exception that you won't be able to output to both an auxiliary out and out to your sound card.

```
bus_config(“SPLITTER”, “aux 0-1 in”, “aux 2-3 out”, “aux 5 out”, “aux 12-13 out”)// OK

bus_config(“SPLITTER”, “aux 0-1 in”, “aux 2 out”,  “aux 7-8 out”, “out 0-1”)// NOT OK
```

This is a powerful way to control a large number of effects in your score file. Here's an example of this technique in action, which sends FMINST( ) to both a main output and via an aux send to SHAPE( ).

```
rtsetparams(44100, 2)
load("FMINST")
load("SHAPE")
load("SPLITTER")
srand()

// This is how we'll route our sounds
bus_config("FMINST", "aux 0-1 out")

bus_config("SPLITTER", "aux 0-1 in", "out 0-1")
```

```
SPLITTER(start = 0, instart = 0, duration = 60, amplitude = 1.0, input_channel = 0,
amp0 = 1.0, amp1 = 1.0)


bus_config("SPLITTER", "aux 0-1 in", "aux 2-3 out")
SPLITTER(start, instart, duration, amplitude, input_channel, amp0, amp1)

bus_config("SHAPE", "aux 2-3 in", "out 0-1")

//-------------------------FMINST
amplitude = 3000
envelope = maketable("curve", 1000, 0,0,0, 0.2,0.5,10.0, 0.7,0.01,10.0, 0.8,0)
carrier = 7.02
modulator = maketable("random", 3, "even", 440, 550)
min_index = trand(1,10)
max_index = 20 - min_index
pan = makeLFO("sine", 10.125, 1,0)
waveform = maketable("wave", 8, "sine")
guide = maketable("cheby", "nonorm", 10, 0.9, 1.0, 1.0, 0.3, -0.2, 0.6, -0.7, 0.9,
              -0.1, 0.0, -0.25, 1.0, 0.25, -0.125)

FMINST(start, duration, amplitude*envelope, cpspch(carrier), modulator, min_index,
max_index, pan, waveform, guide)

//-------------------------SHAPE
min_distortion = 0.5
max_distortion = 1.0
normalization = 0
input_channel = 0
pan = makeLFO("sine", 0.125, 0,1)
transfer_function = maketable("cheby", "nonorm", 1000, 0.4, -0.25, 0.7, 0.6, 0.7, 0.9)

SHAPE(start = 0, instart = 0, duration, amplitude = 1.0, min_distortion,
max_distortion, normalization, input_channel, pan, transfer_function)
```

Score file 29: Multiple outputs with SPLITTER( )

You'll note that both FMINST( ) and SHAPE( ) are utilizing makeLFO( ) for panning, albeit with one moving faster than the other. This is to highlight the

fact that SPLITTER( ) is doing its job and not only taking the frequency modulation sounds directly to our main output, but also sending them for further sculpting. SHAPE( ) is a waveshaping instrument, which is an example of distortion synthesis that utilizes a so-called nonlinear transfer function to distort the incoming signal. This will reshape the incoming signal by adding the presence of new frequencies to its spectrum. One of my favorite transfer functions uses Chebyshev polynomials, which are defined by Forman S. Acton is his book *Numerical Methods that Work* as "cosine curves with a somewhat disturbed horizontal scale, but the vertical scale has not been touched."

In the case of maketable("cheby") we're stating that we don't want the values of our table to be normalized,

Recall score file 17 for a moment. When you heard it, you probably remember that because we called rtsetparams(44100, 1) at the top of our score file — due to the fact that we were working with mono audio files — we only heard playback in one of our speakers. Getting that mono input to output to stereo is another useful strategy for SPLITTER( ). At the bottom of the loop in score file 17, simply add the following after line 49 (after, of course ensuring that we've called on rtsetparams( ) for stereo output).

```
bus_config("DELAY", "in 0", "aux 0 out")
DELAY(start, instart = 0, duration = DUR(), amplitude = 1.0, delaytime = DUR())

bus_config("SPLITTER", "aux 0 in", "out 0-1")
SPLITTER(start, instart, duration, amplitude, input_channel = 0, amp0 = 1.0, amp1 = 1.0)
```

Now let's go back to using an audio sound sample and we'll then pass it to a variety of subtractive synthesis instruments, or filters.

One of the more important aspects of audio sculpting in electroacoustic music is the use of digital audio filters and equalization to better refine the amplitudes of frequencies across the audio spectrum in our compositions. RTcmix boasts a large number of instruments for successfully realizing filtering in your work and we'll start with the basics.

We're going to spend a lot more time taking a look at subtractive synthesis in Interlude V shortly.

Of the many varieties of digital audio filters, there are two whose names are pretty self-explanatory: Low pass filters will allow frequencies to pass below a specified cutoff frequency and high pass filters will do the same, albeit passing frequencies above the cutoff. The signal that is allowed to pass is called the passband and the signal that is filtered out is called the stopband. Those of you familiar with Pure Data will probably be thinking about [lop~] and [hip~] objects and how they are used, however RTcmix doesn't differentiate between low and high pass filters as separate instruments, per se, so instead we'll need to define specific parameters in one instrument to get the job done.

The first filter instrument that we'll explore is ELL( ), which is an elliptical filter. Elliptical filters have sharply defined passbands and stop bands (fast cutoffs), with a certain amount of "ripple" present at each band. This means that the cutoff frequency that you declare will be pretty well defined and steep in comparison to some other filters that tend to rolloff a bit more.

Right now, we're at one of those points where this could easily delve more into a detailed description of fil-

In the following score file, we'll take an input sound and send the left channel to a low pass filter and the right to a high pass filter with randomly derived cutoff frequencies.

Graphs for elliptical filters look like vertical walls while others look like gently rolling hills. There are many

```
rtsetparams(44100, 2)
load("STEREO")
load("ELL")

rtinput("/Users/jerod_s/Desktop/bwv20.aif")

//some global variables
start = 0
instart = 0
duration = DUR()
amplitude = 1.0

//read in our audio file and send to an aux line
bus_config("STEREO", "in 0-1", "aux 0-1 out")
STEREO(start, instart, duration, amplitude, pan = 0.5)

// lowpass filter, creating a stopband at around 100 Hz
passbandcutoff = 499 //in Hz
stopbandcutoff = 501
p2 = 0 // this p-field is always 0 for low and highpass filters
ripple = .8 // amount of ripple (0.0 - 1.0)
attenuation = 90 // in dB (higher the number, the steeper the filter)
ellset(passbandcutoff, stopbandcutoff, p2, ripple, attenuation)

ringduration = 1.0 // in seconds (a slight ringing for sound to fade out)

bus_config("ELL", "aux 0 in", "out 0")
ELL(start, instart, duration, amplitude, ringduration)

// highpass filter, creating a stopband at around 1000 Hz
passbandcutoff = 1001
stopbandcutoff = 999
p2 = 0
ripple = .8
attenuation = 90
ellset(passbandcutoff, stopbandcutoff, p2, ripple, attenuation)

bus_config("ELL", "aux 1 in", "out 1")
ELL(start, instart, duration, amplitude = 2.0, ringduration)
```

Score file 30: Low and high pass filters

Careful copiers of the above code will hopefully get to the last line and notice that I've asked for an amplitude value of 2.0! This will happen from time to time in your score files that utilize filtering because we've filtered out quite a bit of the audio spectrum, so there'll be times like this where we might need to cook the amplitude to compensate for what could be a pretty quiet moment your audio file.

The two most important aspects of ELL( ) and its design are the p-fields for the passband and stopband, which go to a subcommand that is unique to ELL( ) called ellset( ). Note that in order to use a lowpass filter, the p-field for the stopband's cutoff frequency needs to be higher than that of the passband and vice versa for creating a highpass filter. Moreover, these p-fields can be controlled using maketable( ) in order to sweep along different frequencies in real time.

> ELL( ) will also be able to generate a bandpass filter, which has some variation in its p-fields for ellset( ) and des-

At this point, we have two separate, pretty clearly demarcated ideas going on. During the composition process, you might be asking yourself, "But what more can I do with this? It's a good idea to split the high and low pass filtered sounds between the different stereo channels, but what if I wanted to move those around a bit? Could each separate sound pan in an interesting way, more or less interacting with the other sound?"

The answer to all of this or really any ideas you come up with during your composing time is of course, yes, but it's just a matter of identifying the problem at hand and getting a grasp on how to implement it using the program at your disposal. Since bus_config( ) is pretty limitless, we can continue sending our

sounds out of it to more aux lines and we can introduce the PAN( ) instrument to realize our goal here.

> You'd have to have a pretty crazy amount of aux lines going to really test the limits of bus_config( ).

```
rtsetparams(44100, 2)
load("STEREO")
load("PAN")
load("ELL")

rtinput("/Users/jerod_s/Desktop/bwv20.aif")

start = 0
instart = 0
duration = DUR()
amplitude = 1.0

bus_config("STEREO", "in 0-1", "aux 0-1 out")
STEREO(start, instart, duration, amplitude, pan = 0.5)

passbandcutoff = 499
stopbandcutoff = 501
p2 = 0
ripple = .8
attenuation = 90
ellset(passbandcutoff, stopbandcutoff, p2, ripple, attenuation)

ringduration = 1.0

bus_config("ELL", "aux 0 in", "aux 2 out")
inputchannel = 0
ELL(start, instart, duration, amplitude, ringduration)

pan1 = maketable("line", 10, 0,0, 0.5,1, 1.0,0)
bus_config("PAN", "aux 2 in", "out 0-1")
PAN(start, instart, duration, amplitude, inputchannel, panmode = 1, pan1)

passbandcutoff = 1001
stopbandcutoff = 999
p2 = 0
ripple = .8
attenuation = 90
```

```
ellset(passbandcutoff, stopbandcutoff, p2, ripple, attenuation)

bus_config("ELL", "aux 1 in", "aux 3 out")
ELL(start, instart, duration, amplitude = 2.0, ringduration)

bus_config("PAN", "aux 3 in", "out 0-1")
pan2 = maketable("line", 10, 0,1, 0.5,0, 1.0,0)
PAN(start, instart, duration, amplitude, inputchannel, panmode = 0, pan2)
```

Score file 31: Sweepable panning

In my experience, I've found that using PAN( ) as a final instrument call in my score files will ensure that I'm completely and totally in control of how I'm shaping and hearing my sounds in the designated stereo space.

It was alluded to in an earlier footnote, but there are a multitude of instruments that are capable for realizing surround sound directly in RTcmix. That isn't to say, however, that you won't be able to export your sounds as audio files using rtoutput( ) to bring into your digital audio workstation of choice for not only further refinement, but also routing to you surround mix.

```
rtoutput("/path/to/your/file.aiff")
```

Don't forget that simply writing rtoutput("myfile.aiff") will save the file to your current directory. Tired of

If you are committed to rerunning your score file that includes saving to an audio file via rtoutput( ), you'll need to call on the command set_option( ).

```
rtoutput("/path/to/your/file.wav")
set_option("clobber = on") // "clobber = on" will overwrite files.
```

There are a number of RTcmix options that can be turned on or off using set_option( ) and the online documentation will walk you through all of them. Suffice it to say that there are quite a few ways to interact with RTcmix and you are free to customize that experience as much as you'd like to. For example, if I'm running a score with quite a few iterations through my loops or nested loops, I'll often use the command print_off( ) to prevent the long sequence of commands and parameters that I'm running from printing to the Terminal. This will keep your Terminal window clean, but won't save any time for score file rendering, unfortunately.

> But consider what it must've been like before the days of real time synthesis when you'd have to wait a day

With a wide array of instruments at its disposal, RTcmix won't disappoint when it comes to completely customizing and designing sound. With a myriad of ways to internally send and receive digitals via bus_config( ) and SPLITTER( ), you'll be sure to find yourself incorporating subtle changes to your sounds to better enhance your compositional experience. Who knows, maybe you'll find yourself feeling more apt to algorithmically design a sweepable EQ or completely mix your work in a series of .sco files?

Interlude IV: Complex Panning Etude

Using bus_config( ) and SPLITTER( ), create an etude that focuses nearly exclusively on panning. Think about ways that you can isolate sounds exclusively to either side of the stereo field and have them interact in a meaningful way. Could you create a panning etude that has two sounds constantly crossing over each other at different rates, while maintaining their unimpeded signal in each's respective beginning location?

|| Sonata V: Using RTcmix the "wrong" way ||

Of course there is no "right" or "wrong" way to use RTcmix and the title of this Sonata shouldn't instill a feeling that your personal use of the program will inherently be thought of as either par for the course or unusual in any respect. However, for the past few years, I've become increasingly interested in a sub-genre of electronic music referred to as "glitch" and the ways that I can achieve those glitch sounds and textures into my own music using RTcmix. Thus, what follows is a creative exploration of glitch techniques using RTcmix that in no way "breaks" the program or suggests that we'll use it in any way that is unintended. In fact, for me, half of the fun of glitch is using what are (were?) at one time or another thought of as mistakes in the digital audio domain (think clicks and pops and distortion, etc.) that can be creatively harnessed into an exciting sound world.

Glitch music itself could mean a variety of things, including the suggestion that the music itself is borne from some form of failure or destruction. One example is the systematic marring of CDs with the intention that they will skip (codified by, among others, Oval and Yasunao Tone), providing a rhythmic backdrop of unintentional jumps and clicks. Anyone who has plugged an 1/8" jack into their laptop while connected to a mixer with

A definitive look at the history and discussion of glitch music is undoubtedly Caleb Kelly's *Cracked Media:*

the levels up will certainly understand the wonderfully egregious sounds (glitches) that emanate from your studio monitors.

Generating glitch sounds don't always need to come from alterations or cracks in hardware however, and can be cultivated systematically from digital signal processing using RTcmix. This Sonata uncovers a few of those processes and hopefully engenders a willingness to explore the program beyond its perceived functionality. We'll begin by looking at quantization noise and aliasing — which will inevitably lead to a brief discussion of digital audio theory — and move to manufacturing clicks and pops, finally exploring ways to call on RTcmix instruments beyond their "limits."

As mentioned earlier, bit depth is the number of bits available to represent each sample in the digital domain. In 16 bit audio, there are 65,536 integer values available to represent the analog signal being sampled (with 44,100 samples/second in standard CD quality audio). However, analog signals will always be more accurate than their digital representations and quantization error is the resulting numerical difference between incoming, continuous analog signals and their discrete digital representations during the sampling process. This inherently produces noise that will become far more apparent as the bit depth is lowered.

> Although introducing a small amount of noise (called dithering) into the sampling process randomizes the

Introducing quantization noise is a simple procedure in RTcmix using the DECIMATE() instrument.

```
rtsetparams(44100, 2)
load("DECIMATE")
rtinput("/path/to/your/file.aiff")

inputstart = 0
duration = 3.0
```

```
preamplitude = 0.5   //amplitude of original signal
postamplitude = 0.5  //amplitude of decimated signal
bitdepth = 16.0
filtercut = 0
inputchannel = 0
pan = 0.5

for(start = 0; start < 40; start += duration){
     bitdepth -= 2.0    //decrease each time through the loop
     DECIMATE(start, inputstart, duration, preamplitude, postamplitude,
     bitdepth, inputchannel, pan)
     }
```

Score file 32: Quantization error

For this score file, the bit depth is initialized at 16.0 and reduced by 2.0 each time through our loop. What results is a gradual shift from the clean sound of the original sampled file to something far more noisy. I've found that when sampling crisp sounds from something like a music box will turn its tones into something akin to an electric guitar with feedback, for example. There are many ways to refine your decimated sounds, including experimenting with the amplitude values for the incoming and decimated signals, as well as going so low as to recreate 1-bit sounds in a simple on/off scenario, which is great for those who have an affinity for harsh noise.

Now, whether or not you want to cook your amplitudes above 1.0 for some extra shock value (or Merzbow

According to the Nyquist theorem, frequencies that can be accurately sampled in the digital domain are less than or equal to one half of the sample rate. Thus, when sampling frequencies at 44.1 kHz, those frequencies greater than 22.05

The general, agreed upon range of human hearing is about 20 Hz to 20 kHz, so the Nyquist frequency is

kHz will be aliased, or folded over, which produces inaccurate, noisy representations in the range of frequencies that we can hear.

The following equation is used to find the frequency of the aliased signal in Hertz. In the score file example that follows, a sine wave is sampled at 39.1 kHz, which when folded over produces a tone at 5,000 Hz.

$$a(N) = |s - (N)r|$$

a = frequency of aliased signal in Hz
s = frequency of sampled signal in Hz
r = sample rate
N = nearest common multiple (integer) of a and r

$$a(1) = |39.1 - (1)44.1|$$
$$a = |\text{-}5|$$
$$a = 5 \text{ kHz}$$

```
rtsetparams(44100, 2)
load(“WAVETABLE”)

start = 0
duration = 10
amplitude = 5000
envelope = maketable(“line”, 1000, 0,0, 0.2,1, 0.4,0.5, 0.7,0.5, 1,0)
frequency = 39100  // some frequency > 22.05 kHz (Nyquist frequency)
pan = 0.5
waveform = maketable(“wave”, 1000, “sine”) // sine wave
WAVETABLE(start, duration, amplitude*envelope, frequency, pan, waveform)
```

Score file 33: Aliased signal

Outside of the glaring initialization of such an outstanding supersonic frequency, the aural result of this score file fails to provide anything unusual. However, aliased score files that utilize wave forms with more robust overtone content that shift frequencies within a loop structure provide more interesting results. The following example uses sawtooth waves with a random walk algorithm to change frequencies over time.

```
rtsetparams(44100, 2)
load("WAVETABLE")

srand()

envelope = maketable("line", 1000, 0,0, 0.5,1.0, 1.0,0)
frequency = 37100
pan = 0.5
waveform = maketable("wave", 1000, "saw")

for(start = 0; start < 40; start += 1){
     walkvalue = irand(0, 1) // generate random number between 0 and 1

             if(walkvalue <= 0.7){   // decrease by 0.5 Hz freq 70 % of time
                     frequency -= 20.5
                 }

            else{                          // else increase by 0.5 Hz 30 % of time
                     frequency += 20.5
                }

     duration = irand(2, 10)
    amplitude = irand(500, 4000)
     WAVETABLE(start, duration, amplitude*envelope, frequency, pan,
     waveform)
    }
```

Score file 34: Aliasing with random walk

The snaps, clicks, and pops of speaker cones aided by sudden, drastic shifts of amplitude are some of the more salient features of glitch. Once considered the unwanted byproducts of the mastering and mixing process — and one of the closest notions of a "mistake" when working with digital audio that strives for clean signals — these sounds have been successfully implemented by many artists.

Using WAVETABLE() will allow us to specify a wave shape with a specific duration, amplitude, and frequency, as well as a custom waveform. As we are now well aware, maketable( ) is able to generate any of a number of graphs, and to generate clicks, we'll use a series of line segments denoted by "line", although "curve" and other shapes can be specified followed by the number of points in the graph. Again, I like to represent the specific values (x, y) of the graph in pairs with no spaces.

```
rtsetparams(44100, 2)
load("WAVETABLE")
```

```
duration = 1 // total duration
amplitude = 30000
frequency = 2  // initial frequency
waveform = maketable("line", 32767, 0,0, 16384,1, 16385,-1, 16386,0, 32767,0)

for(start = 0; start < 20; start += 1){
     // random #s between 0 and 0.5, avoid values > 3
     frequency = irand(0, 0.5)
     pan = irand(0, 1)

    WAVETABLE(start, duration, amplitude, frequency, pan, waveform)
    }
```

Score file 35: Generating clicks from maketable( )

Creating a table with 32,767 points in straight line segments corresponds directly with 2^{16} samples in 16-bit audio. Using values less than 32,767 table points will begin to take on a discernible pitch as the distance between each respective point becomes larger. Moreover, specifying frequency values greater than three should be avoided, as the pop will cycle through our for( ) loop fast enough to coalesce into a discernible pitch. This particular maketable( ) writes the pop directly in the middle of the graph, going from the amplitudes 1 to -1 and back to 0. Shifting the frequency values will change the rate at which the pops occur, though using irand( ) also gives this score much more of a "popcorn" effect, as it returns random floating-point values between 0 and 1 each time through the loop. For a more interesting texture, panning has also been randomly distributed across the stereo field.

These particular clicks are intrinsically harsh: Their function is to send the speaker cone from full compression to complete rarefaction in a nearly indiscernible amount of time. Playing them back at loud amplitudes over long periods of time can be damaging to the speakers and the user. However, raw click sounds can be molded into something much more subtle and beautiful using filters.

```
rtsetparams(44100, 2)
load("WAVETABLE")
load("ELL")

bus_config("WAVETABLE", "aux 0-1 out”)
bus_config("ELL", "aux 0-1 in", "out 0-1")
srand()
duration = 1
amplitude = 8000
frequency = 2
waveform = maketable("line", 32767, 0,0, 16384,1, 16385,-1, 16386,0, 32767,0)

pbcut = 9000      // passband cutoff frequency in Hz
sbcut = 900       // stopband cutoff frequency in Hz
ripple = 0.2       // amount of ripple (dB)
```

```
attenuation = 90  // attenuation at stopband (dB)

ellamp = 9        // filter amplitude
ringdur = .8      // ring-down duration (in sec.)

// passband value > stopband = highpass filter
ellset(pbcut, sbcut, 0, ripple, attenuation)

for(start = 0; start < 20; start += 1){
     frequency = irand(0, 0.5)
     pan = irand(0, 1)
     WAVETABLE(start, duration, amplitude, frequency, 0.5, waveform)
     ELL(start, 0, duration, ellamp, ringdur, 0, pan)
     }
```

Score file 36: Filtering clicks with ELL( )

We will again call on ELL( ), our elliptical filter, which is customizable by way of its p-fields, and can function as either a high pass, low pass, or band pass filter. To review, ELL( ) first requires the ellset( ) command to specify its parameters before being called in the score file. This score file uses a high pass filter to accentuate the sharp, snapping quality of the clicks. To achieve this, the passband cutoff frequency (9,000 Hz) must be greater than the stop band cutoff (900 Hz). The amount of ripple, or ringing in the filter, is specified in dB. In this instance, larger values such as 50 dB will transform the clicks into a discernible pitch, thus the very low value of 0.2 dB. Finally, the amount of attenuation at the stop band is set to 90 dB, a steep filter that works well for this intended effect.

> As long as we're on the subject of using ELL( ) in our score files, I'd like to point out that culminating your

After filling out the parameters for ellset( ), a few P-field commands need to be specified for ELL( ) itself. Outside of the input and output start times, the filter's duration — set to the same value of WAVETABLE( ) — amplitude and

ring-down duration need to be declared either as numerical values or again in this instance as variables. Although panning was set to the center in WAVETABLE( ), it was again randomized within the loop for ELL( ).

Panning plays a key role in another useful approach for creating clicks and pops. Rather than setting the pan value directly in the middle of the stereo field — or even a random distribution specified with irand( ) — it is possible to control spatialization through the makeLFO( ) command.

```
rtsetparams(44100, 2)
load("WAVETABLE")

duration = 5
amplitude = 2000
envelope = maketable("curve", 1000, 0,0,1, 1,1,0, 3,1,1, 4,0)
frequency = 100
pan = makeLFO("sine", 5.0, 0.0, 1.0)
waveform = maketable("wave", 1000, "sine")

for(start = 0; start < 20; start += 1){
     WAVETABLE(start, duration, amplitude*envelope, frequency, pan, waveform)
    }
```

Score file 37: makeLFO( ) within loops for clicks

The frequency for makeLFO( ) is set at a 5.0 Hz sine wave, slowly oscillating between 1.0 (stereo left) and 0.0 (stereo right). By using the makeLFO( ) as a variable for panning and then putting it in the loop structure, clicks will occur as the pan values continually reset. One will notice that by simply eliminating the loop and initiating the WAVETABLE() instrument with a start time of 0 (substitute start with 0), no clicks will materialize and the score will produce a 100 Hz sine wave for five seconds. Moreover, the frequency values in makeLFO() can be

altered to produce interesting effects. Frequencies ranging from 0.1 Hz to 20 Hz will simply produce clicks and pops against the frequency of the sine wave specified in the MAKETABLE() instrument, while anything greater than 20 Hz will begin to interact with the sine wave, producing a sound akin to ring modulation.

The following score file features a series of overlapping and unfolding triangle and sine waves with aliased frequencies. In its various transformations, this score file produced much of the music in in my electroacoustic work *kernel_panic* from about 4'25" to 6'30".

```
rtsetparams(44100, 2)
load("WAVETABLE")
load("ELL")

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("ELL", "aux 0-1 in", "out 0-1")
srand()

pbcut = 80
sbcut = 24000
ripple = .8
attenuation = 90

ellamp = 60
ringdur = .1
dur = 1
amp = 9000
// envelope with curved line segments
env = maketable("curve", 1000, 0,0,1, 1,1,0, 3,1,-1, 4,0)
freq1 = 22000
pan1 = makeLFO("sine", 1, 0, 0.5)

freq2 = 20000
pan2 = makeLFO("sine", 3.5, 0.5, 1)

triwave = maketable("wave", 1000, "tri")
sinewave = maketable("wave", 1000, "sine")
```

```
for (st = 0; st < 100; st += 1){
     freq1 += irand(0, 40)
     freq2 -= irand(0, 40)
     dur = irand(5, 10)
     pbcut += 5
     sbcut -= 10
     st += irand(0, 3)

     if (st > 60){
          freq1 -= irand(0, 40)
          freq2 += irand(0, 40)
          sbcut += 100
          }

// triangle waves produce discernible pitches, sine waves will click
     WAVETABLE(st, dur, amp * env, freq1, pan1, triwave)
     WAVETABLE(st, dur, amp * env, freq2, pan2, triwave)

     WAVETABLE(st, dur, amp * env, freq1, pan1, sinewave)
     WAVETABLE(st, dur, amp * env, freq2, pan2, sinewave)
     }

ellset(pbcut, sbcut, 0, ripple, attenuation)
ELL(start = 0, instart = 0, dur = 100, ellamp, ringdur, 0, 0.5)
// keep dur (p3) as long as total script time
```

kernel_panic,
in its entirety.

Score file 38: Score file from kernel_panic

Using triangle waves in maketable("wave") produces discernible pitches as each subsequent harmonic frequency is folded over. Conversely, the aliased fundamental frequencies from the sine waves produce much softer pitched sounds that enhance the constantly resetting pan values. Moreover, the variables for WAVETABLE( ) and ELL( ) constantly change via the irand( ) command in the loop. Despite being a relatively simple score file, it produces a variety of results as its variables — such as those for frequencies, durations, and filter parameters

— are altered. Using a variety of waveforms and a multitude of envelope shapes further adds to the effect of different waves entering in canon.

The Synthesis Tool Kit (STK) was developed by Perry Cook and Gary Scavone as a means to, among other things, design and port physical modeling instruments to a variety of programs. One famous example of physical modeling is the so-called Karplus-Strong Plucked String Algorithm, which sends bursts of white noise to a feedback loop in a short delay line to replicate the phenomena of plucked strings, such as harpsichords or guitars. In the STK and RTcmix, you can find these plucked string sounds in the form of the STRUM( ) instrument. In fact, we encountered one of Brad Garton's example score files during our installation of the standalone version of RTcmix, but we'll revisit it now in its entirety.

> You'll find this score file and the rest of the STK examples in /path/to/RTcmix/docs/sample_scores. This

```
/* START:
   p0 = start; p1 = dur; p2 = pitch (oct.pc); p3 = fundamental decay time
   p4 = nyquist decay time; p5 = amp, p6 = squish; p7 = stereo spread [optional]
   p8 = flag for deleting pluck arrays (used by FRET, BEND, etc.) [optional]
*/

rtsetparams(44100, 2)
load("STRUM")
makegen(2, 2, 7, 7.00, 7.02, 7.05, 7.07, 7.10, 8.00, 8.07)

srand(0)
for (st = 0; st < 15; st = st + 0.1) {
     pind = random() * 7
     pitch = sampfunc(2, pind)

     START(st, 1.0, pitch, 1.0, 0.1, 10000.0, 1, random())
}
```

> While I of course always advocate for you to scour the documentation on your own, you won't find any refer-

Score file 38: Brad Garton's STRUM1.sco example

I made no changes to the example file to give you an example of one of RTcmix's original author's code, which is a tremendously valuable learning experience for anyone looking to use RTcmix. You'll note that STRUM( ) is being loaded, but was called in our loop as START( ). The STRUM( ) instrument proper is really a family of plucked string instruments, which includes START( ), BEND( ), FRET( ), START1( ), BEND1( ), FRET1( ), VSTART1( ), and VFRET1( ). Each has its own unique take on the plucked string algorithm, and for lack of better words, START( ) is the most basic, though it hasn't been updated for use with maketable( ), hence the call to makegen( ) for its envelope shape.

Take a moment to scour Brad's code and come up with some educated guesses as to what exactly all of the p-fields represent. You probably came up (going from left to right) p-field 0 for a start time, then duration, then a value for pitch, and then a series of values that became ambiguous. P-fields 3 and 4 represent a fundamental delay time and Nyquist delay time, respectively. Following that is amplitude (absolute, of course) and a "squish" time. Lastly, a value for pan is declared, which has been randomized.

> Or, how soft is the item that is "plucking the string?" 0 = super hard while 10 is akin to fleshy pads of fingers.

Now, let's have a little bit of fun with this score file and glitch it out just a bit.

```
rtsetparams(44100, 2)
load("STRUM")

table_number = 2
makegen(table_number, 2, 7, 3.00, 4.02, 7.05, 7.07, 7.10, 8.00, 18.07)

srand()

increment = 0.1
```

```
for (start = 0; start < 10; start += increment) {
     pitch_index = random() * 7
     pitch = sampfunc(table_number, pitch_index)
     amplitude = pickrand(500, 10000, 15000, 20000)
     decay = random()
     nyquist_decay = random()
     squish = start
     START(start, duration = increment, pitch, decay, nyquist_decay, amplitude,
            squish, random())
     }
```

Score file 39: "Glitched" version of score file 38

We need to first make sense of the makegen( ). This particular type of makegen( ) is a type-2, which fills a table with random values, similar to what we've already encountered with maketable("random"). In this makegen( ), we have seven elements, which are octave point pitch class designators for pitches. In the loop, we're calling on each of those elements, randomly, via sampfunc( ) and a variable for the index of our makegen( ) itself, called pitch_index. From there, we've randomized a number of parameters and incremented our squish value to go from a super hard plectra, to the fattest of fingers. None of this, however, makes our score file particularly glitchy, and when you've played it back, you'll notice some clicks, but where do they come from?

Take a closer look at the list of pitches in makegen( ). You'll see that I've altered a few away from the comfortable octave range of 7 and 8 to the extremes, namely 2 and 18. These two octaves fall well outside of the range of frequencies that we could reasonably expect a more "normal" sized string to be able to play (unless we were doing something along the lines of Alvin Lucier's music for long, thin wires), and so these parameters are essentially breaking the algorithm for pluck strings, at least in the strictures set by the code that constitutes the

START( ) version of STRUM( ), and we get resulting clicks in our audio, which again can be further refined or sculpted using filters or delay lines, etc.

The final example score file in this Sonata draws upon many of the aforementioned techniques, but introduces the control_rate( ) command. Each time that you call upon a table in RTcmix, the program will need to essentially "sample" the table in question. This is analogous to the way that digital sampling represent waveforms at 44,100 samples/second, and the standard control sampling rate for RTcmix is 1000 times/second. You are able to sample your control function at the sample rate by using control_rate(44100), but this can be expensive on your CPU and you probably won't notice a tremendous difference in the end. However, lowering the control sample rate can have a deliciously adverse effect on your sounds and can in some instances get you to the lo-fi "8-bit" sound that you've been craving using control_rate(8).

Even though, for example, you may have designed a sine wave using maketable("wave", 1000, "sine"), which

```
rtsetparams(44100, 2)
load("FMINST")
load("WAVETABLE")
control_rate(4)  // have fun experimenting with these values
print_off()
srand()

//----------- FM
start = 0
notedur = 0.05
amplitude = 20000
carrier_frequency = 22051
carrier_envelope = maketable("line", 100, 0,0, 1,0.5, 80,0.5, 90,0, 100,0)
modulator_frequency = (carrier_frequency * (2 / 3))
modulator_envelope = maketable("line", 100, 0,0, 1,1.0, 90,0.3, 100,0)
max_index = 1
min_index = 1
waveform = maketable("wave", 1000, "sine")
index_wave = maketable("window", 1000, "hanning")

iteration = 0.05
```

Although it is reserved specifically in the Perl language, reset( ) will do exactly the same thing as control_rate(

```
for(start = 0; start < 100; start = start + iteration) {
     pan = random()
     FMINST(start, notedur, amplitude,carrier_frequency*carrier_envelope,
     modulator_frequency*modulator_envelope,max_index, min_index, pan, waveform, index_wave)

     x = random()
     y = random()

     max_index = (0.166666 * ((x + y) - 20) + 0.5)
     carrier_frequency *= max_index

     if(x <= 0.5){
          modulator_frequency += irand(1, 3)
          min_index = ((x + y) / 2)
          }

     if(x > 0.9){
          carrier_frequency += 100
          modulator_frequency -= 100
          }
     }

//------------ WAVETABLE
start = 0
duration = 4.5
amplitude = 5000
envelope = maketable("line", 100, 0,1, 0.99,1, 1,0)
frequency = 40
pan = makeLFO("sine", 2.0, 0,1)
waveform = maketable("wave", 1000, "tri")

for(start = 0; start < 35; start += 7){
     WAVETABLE(start, duration, amplitude*envelope, frequency, pan, waveform)
     duration += 0.5
     }
```

Score file 40: Fun with control_rate( )

There are any of a number of ways to creatively explore RTcmix and the best advice for realizing glitches textures in your music is to start with some of the basic tenets of digital signal processing and don't be afraid to stretch the perceived limits of the instrument or command at hand and trust your ears when creatively applying some of these examples into your own work. The Synthesis Tool Kit is one example of a set of instruments that may be exactly what you are looking for in a suite of tools to help you realize some of the more classic sounds of 90s computer music — but like any instruments in RTcmix — can be further enhanced to extend far beyond its original intended use.

Sonata V: Glitch etude

Begin by using an example score file from the RTcmix application's documentation and step-by-step alter it to either distort it or glitch it in some meaningful way. Write down each of the steps that you used to realize your work and present it in a before and after demonstration for your peers.

|| Interlude V: More synthesis and modulation ||

By now, we've either briefly discussed or used the sounds of a variety of types of digital synthesis procedures. This interlude will explore three main tenets of synthesis in a bit more detail: Additive, subtractive, and granular synthesis, while also discussing ring and frequency modulation.

Our very first foray into RTcmix was by way of the innocuous sine wave, which is one of the most basic waveforms in electronic music. Sine waves are devoid of any harmonic content, meaning that they singularly sound at a fundamental pitch — calculated in cycles per second or Hertz (Hz) — and nothing more, which is why they are sometimes referred to as pure waves. However, by adding sine waves above the fundamental pitch together at specified amplitudes and frequencies, it is possible to construct more complex waveforms through additive synthesis.

The harmonic series is a naturally occurring phenomenon that outlines the respective partials (or harmonics) of any fundamental pitch. For example, a string that is tuned to 110 Hz will sound at 220 Hz when it is shortened to half of its length. If you were to half the half, then half the half, etc., you'd find that each respective harmonic sounds at 330 Hz, 440 Hz, 550 Hz, etc. This harmonic series is critical in constructing waveforms from scratch. Some of our most basic waveforms in electronic music are, in addition to the sine wave, square, sawtooth, and triangle waves.

A triangle wave is built by taking a fundamental pitch and adding only the odd numbered partials to its spectrum at amplitudes of $1/n^2$ where n is equal to the partial number. So, we can build a triangle wave using sine waves at 110 Hz (1.0 amplitude), 330 (0.33), 550 (0.2), 770 (0.14), etc. When built this way in a digital audio workstation, you'll find that the more partials that are added, the more discernible your waveform will begin to look.

> This is another wholly vague and glossed over explanation of basic additive synthesis, but again, I encourage

Thankfully, RTcmix doesn't make us do all of the math ourselves when calling for various waveforms via maketable("wave") and there may be some times when you'd like more control over the constructs of your waveform. For this, maketable("wave3") will be your resource, which gives you the ability to specify not only partials and amplitudes, but also phase shifts between 0 and 360 degrees.

> This isn't to say that you couldn't construct, say, a crude triangle wave by using maketable("wave", 1000, 1.0,

```
wave = maketable("wave3", 1000, 1,1,0, 3,0.33,45, 5,0.2,90, 7,0.14,120)
```

This whole business of constructing waves becomes way more fun when you use them in conjunction with HALFWAVE( ). A versatile instrument, HALFWAVE( ) essentially joins two waveforms together with the ability to dynamically update their meeting point. In the following example score file, I'm taking a variety of waveforms and randomly selecting from them to construct the two waves that I'll need for each call to HALFWAVE( ) inside of my loop. The most critical part of this process is accomplished by these two lines of code, which specify the array of waveforms that I'm using as well as a variable for the length of my array, via len( ):

```
wavegamut = {wave1, wave2, wave3, wave4, wave5}
wavegamutlength = len(wavegamut)
```

What follows then, is a sort of nebulous, out of tune organ sound that could be even further refined by finding ways to more creatively design the constituent parts of each call to maketable("wave3").

```
rtsetparams(44100, 2)
load("HALFWAVE")
load("REVERBIT")
srand()

bus_config("HALFWAVE", "aux 0-1 out")
bus_config("REVERBIT", "aux 0-1 in", "out 0-1")

totalduration = 60

//-----------------------HALFWAVE
start = 0
duration = 1.75

octavegamut = {2, 10, 11}
octavegamut_length = len(octavegamut)

pitchgamut = {0.00, 0.01, 0.04, 0.05, 0.06, 0.08, 0.10, 0.11}
pitchgamut_length = len(pitchgamut)

amplitude = 1000
envelope = maketable("line", 100, 0,0, 0.5,1, 1.0,0)
wave1 = maketable("wave3", 1000, 3.14,1,0, 6.28,1,0.5)
wave2 = maketable("wave3", 1000, 1.00,1,0, 2.00,1,0.5)
wave3 = maketable("wave3", 1000, 1,1,0, 3,0.3,0, 5,0.2,0, 7,0.05,0, 9,0.01,0,
                  11,0.001,0)
wave4 = maketable("wave3", 1000, 1,1,0, 2,0.5,0, 3,0.3,0, 4,0.25,0, 5,0.2,0,
6,0.16,0, 7,0.14,0, 8,0.125,0)
wave5 = maketable("wave3", 1000, 1,1,0, 3,0.14,0, 5,0.04,0, 7,0.02,0, 9,0.012,0,
11,0.008,0)
wavegamut = {wave1, wave2, wave3, wave4, wave5}
wavegamutlength = len(wavegamut)
```

```
wavecrossoverpoint = 0.5
pan = 0.5

increment = 0.5

for(start = 0; start < totalduration; start += increment){
     octave = octavegamut[trand(0,octavegamut_length)]
     nextpitch = pitchgamut[trand(0,pitchgamut_length)]
     pitch = nextpitch + octave

     if(octave == 2){
          duration = 5.0
          pan = makeLFO("sine", 1.25, 0,1)
          HALFWAVE(start, duration, cpspch(pitch)+0.04, amplitude*envelope, wave1,
          wave2, wavecrossoverpoint, pan)
          }

     wave1 = wavegamut[trand(0,wavegamutlength)]
     wave2 = wavegamut[trand(0,wavegamutlength)]

     random_number_x = round(trand(0, 10))
     random_number_y = round(trand(0, 10))
     average_of_random_numbers = ((random_number_x + random_number_y) / 2)
     wavecrossoverpoint = average_of_random_numbers / 10

     pan = random()
     HALFWAVE(start, duration, cpspch(pitch), amplitude*envelope, wave1, wave2,
     wavecrossoverpoint, pan)
     increment = average_of_random_numbers / 7
     }

//-----------------------REVERBIT
start = 0
instart = 0
duration = totalduration
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, 0.4,1.0, 0.9,0, 1.0,0)
revtime = maketable("line", 1000, 0,1.0, 0.5,1.0, 1.0,0.2) //keep these short
revamnt = 1.0        //0-1 (dry to wet)
chandelay = maketable("random", 100, "gaussian", 0.01,2.0)
```

```
cutoff = 2000       //low pass filter in Hz
REVERBIT(start, instart, duration, amplitude*envelope, revtime, revamnt, chandelay,
cutoff)
```

Score file 41: HALFWAVE( )

Earlier, we encountered subtractive synthesis using elliptical filters, designing them as either high or low pass, though we also know that ELL( ) can set up bandpass filters as well. If we consider, by way of an example, a high pass filter, we know that its design specifies a cutoff frequency (Hz) and allows frequencies above that cutoff, in the passband, to sound while rejecting those that fall below it in the stopband. Recall also that filters have an amount of ripple at the the meeting point of the pass and stop bands, which is calculated in dB, and is the ratio of the highest and lowest amplitudes in the passband. Perfectly flat ripple would have a gain of 0 dB, while increasing the ratio will start to give your filter a ripply, wavy look to it.

A great explanation of this — and filters in general — is given in Miller Puckette's *The Theory and Technique*

Another versatile filter in RTcmix's arsenal is BUTTER( ), a dynamic Butterworth filter.

```
rtsetparams(44100, 2)
load("TRANS")
load("BUTTER")
rtinput(“/path/to/file.aif”)
control_rate(8)

bus_config("TRANS", "in 0-1", "aux 0-1 out")
bus_config("BUTTER", "aux 0-1 in", "out 0-1")
```

```
//-----------TRANS
start = 0
instart = trand(0,DUR())
duration = 20
amplitude = 1.0
low = octpch(-0.01) //down a m2
high = octpch(0.01) // up a m2
transp = maketable("random", 1000, "gaussian", low,high)
transposition = makeconverter(transp, "pchoct") //convert to oct.pc
TRANS3(start, instart, duration, amplitude, transposition)

//------------FILTER

filt_type = "bandpass"
sharpness = 5
balance = 1
input_channel = 0
pan = 0.5
bypass = 0  // 0 to filter sound, 1 to bypass
center_frequency = 1500
bandwidth = 250
BUTTER(start, instart = 0, duration, amplitude, filt_type, sharpness, balance,
input_channel, pan, bypass, center_frequency, bandwidth)
```

Score file 42: Butterworth bandpass filter

After having a bit of fun with some random transpositions and grittiness via control_rate( ), our Butterworth filter has been initialized as a bandpass, with a center frequency of 1500 Hz and a bandwidth of 250 Hz. Bandwidth measures the width of the passband, again in Hz. This p-field can by dynamically updated and only needs to be included if you are using a bandpass or bandreject filter.

> Head to this Wikipedia link: https://en.wikipedia.org/wiki/Butterworth_filter for a really great visual

When it comes to total refinement of your sound, there really is no better solution than careful equalization, which isn't necessarily a function of filtering out

certain frequencies. Rather, equalization gives you the ability to attenuate a number of frequencies in the audio spectrum, thus enhancing or quieting certain bands of sound that will better direct you to your desired end result and RTcmix's MULTEQ( ) instrument will allow you to attenuate your sounds up to eight bands.

```
rtsetparams(44100, 2)
load("MULTEQ")

rtinput("/path/to/file.aif")

start = 0
instart = 0
duration = DUR()
amplitude = 0.25
bypass = 0

type1 = "lowshelf"
freq1 = makeLFO("tri", 2.25, 20,100)
Q1 = maketable("line", "nonorm", 1000, 0,10, 0.5,1, 1.0,10)
gain1 = maketable("line", "nonorm", 1000, 0,5, 0.25,-10, 1.0,-2)
bypass1 = 0

type2 = "peaknotch"
freq2 = makeLFO("sine", 0.0125, 200,12000)
Q2 = 5
gain2 = maketable("curve", 25, 0,-12,0, 0.25,9,-10, 1.0,0)
bypass2 = 0

type3 = "peaknotch"
freq3 = 15000
Q3 = 0.25
gain3 = maketable("line", "nonorm", 1000, 0,-2, 1.0,12)
bypass3 = 0
MULTEQ(start, instart, duration, amplitude, bypass,
        type1, freq1, Q1, gain1, bypass1,
        type2, freq2, Q2, gain2, bypass2,
        type3, freq3, Q3, gain3, bypass3)
```

Score file 43: MULTEQ( ) with p-field updates

Infinite impulse response (IIR) and finite impulse response (FIR) filters have an output that is dependent on a series of incoming impulses. IIR filters are infinite because they are recursive and have some amount of feedback in their design, while FIR filters are non-recursive. RTcmix will give you the ability to design very complex filters using its IIR( ), FIR( ), or FILTERBANK( ) instruments, but one exciting effect that falls in line with these principles is the comb filter, accessed by way of COMBIT( ) or MULTICOMB( ).

```
rtsetparams(44100, 2)
load("STEREO")
load("COMBIT")
rtinput(“/path/to/file.aif”)

bus_config("STEREO", "in 0-1", "aux 0-1 out")
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
pan = 0.5
STEREO(start, instart, duration, amplitude, pan)

bus_config("COMBIT", "aux 0-1 in", "out 0-1")
frequency = maketable("line", "nonorm", 10, 0,0.1, 1.0,100.0) // in Hz
reverbtime = maketable("line", "nonorm", 1000, 0,2.0, 0.2,10.0, 0.75,4.25,
1.0,0.125)
inputchannel = 0
pan = 0.5
ringduration = 2.5
COMBIT(start, instart, duration, amplitude - 0.25, frequency, reverbtime,
inputchannel, pan, ringduration)
```

Score file 44: Comb filter with linear frequency growth

You'll note that as this score file plays out, the frequency of the comb filter slowly increments from 1 to 100 Hz over a series of varying reverb times. The result is an ethereal cascade of sounds as the comb filter rings the incoming sound at the varying frequencies outlined in our maketable( ) for p-field 4. When viewed on a graph, you'll see why comb filters are aptly named.

There are a number of filtering and equalizing instruments in RTcmix's rich array of tools. As with most sections in this book, it is my hope that your interest will be piqued enough to head to the documentation and for yourself check out the possibilities afforded to you by FILTERBANK( ), FILTSWEEP( ), JFIR( ), MULTICOMB( ), MULTEQ( ), and others.

While it has come to mean a few different things in digital music, sampling is an important component of our work in RTcmix. When we construct a waveform using maketable( ) and WAVETABLE( ) we're creating a series of samples that in sum create a wavetable. There are many different procedures that we can realize using wavetables and when we divide them into several microsonic subparts and play them back rapidly, we are in the realm of granular synthesis.

"Microsonic" is an intentional nod to the work of Curtis Roads, whose book *Microsound* is required reading for

Many composers have explored granular synthesis in one form or another and RTcmix boats a number of instruments that can realize this technique. Here is an introductory example using John Gibson's JGRAN( ) instrument, which defines small grains using a specified waveform.

```
rtsetparams(44100, 2)
```

```
load("JGRAN")
control_rate(512)

start = 0
duration = 10
amplitude = 5.0  //okay to go above 1.0 for this instrument
seed = srand()

//oscillator type: 0 = wavetable, 1 = FM
type = 1

//randomize oscillator phase? 0 = no, 1= yes
randphase = 1

// grain envelope
genv = maketable("window", 1000, "hanning") // common window function

// grain waveform
gwave = maketable("wave", 1000, "sine")

// modulation frequency multiplier
mfreqmult = maketable("line", "nonorm", 25, 0.0,1.25, 0.5,2.0, 1.0,0.5) // Hz

// index of modulation envelope (per grain)
modindex = maketable("line", "nonorm", 100, 0.0,0.5, 0.33,10.0, 0.66,25.0,
0.99,0.0, 1.0,0)

// grain frequency
minfreq = 100
maxfreq = 2205

// grain speed
minspeed = 10    //number of grains/second
maxspeed = 1000

// grain intensity (decibels above 0)
mindb = 40
maxdb = 90

// grain density
density = maketable("line", "nonorm", 100, 0.0,1.0, 0.25,0.325, 1.0,0.8)
```

```
pan = maketable("random", 100, "gaussian", 0,1)

// grain stereo location randomization, 0 = none, 1 = full randomization
panrand = 1

JGRAN(start, duration, amplitude, seed, type, randphase,
      genv, gwave, mfreqmult, modindex, minfreq, maxfreq, minspeed, maxspeed,
      mindb, maxdb, density, pan, panrand)
```

Score file 45: Adapted from John Gibson's JGRAN( ) help file

Note that we have quite a bit of control of how these grains of sound interact with each other over time. Not only can we specify the type of waveform to cull these grains from — in this case a sine wave — but we can specify an envelope shape for them, their frequencies, their speed in grains/second, density, intensity, and stereo location.

The so-called Hanning and Hamming window functions are pretty typical for grain shapes in granular synthesis.

All of these p-fields are tightly controlled and can be modified in real-time via maketable( ). For even greater stochastic control of these grains, Mara Helmuth's SGRANR( ) is a fantastic instrument and we can perform granular synthesis on incoming sound files using GRANULATE( ).

Granular synthesis was a heavily explored synthesis technique as RTcmix came of age, so you'll find quite a few

```
rtsetparams(44100, 2)
load("GRANULATE")

source = "/path/to/file.aif"
file = maketable("soundfile", "nonorm", 0, source)
filedur = DUR()

start = 0
instart = trand(0,DUR())
```

```
duration = 45
amplitude = ampdb(40)
channels = 2
inputchannel = 0

windowstart = 0.0
windowend = random()
wraparound = 1
traverserate = 1.0
table = maketable("window", 1000, "hamming")
hoptime = 0.0625

injitter = 0.0625
outjitter = 0.0125

mindur = hoptime + random()
maxdur = mindur() + random()
minamp = 0.5
maxamp = 2.0

GRANULATE(start, instart, duration, amplitude, file, channels, inputchannel,
windowstart, windowend, wraparound,traverserate, table, hoptime,
injitter, outjitter, mindur, maxdur, minamp ,maxamp)
```

Score file 46: GRANULATE( )

Ring modulation is a classic effect in the analog electronic music world and one of my absolute favorite works that uses the technique is Karlheinz Stockhausen's *Mantra*. Ring modulation involves the multiplication of two signals and in the case of Stockhausen's work, we take the complex sound of a piano and multiply it by the simple signal of a sine wave. This technique can be replicated using the AM( ) instrument.

```
rtsetparams(44100, 2)
load("AM")
rtinput("/path/to/my/awesomefile.wav")
```

```
start = 0
instart = 0
duration = 30
amplitude = 1.0
frequency = makeLFO("sine", 0.1, 30,1000)
inputchannel = 0
pan = 0.5
waveform = maketable("wave", 1000, "sine")
AM(start, instart, duration, amplitude, frequency, inputchannel, pan, waveform)
```

Score file 47: Ring modulation using AM( )

When applied to an incoming signal, ring modulation produces sidebands, which are the sum and difference (in Hz) of the signal corresponding to the frequency of the modulator. For example, a 400 Hz tone that is sent to a 100 Hz ring modulator will produce sidebands of 500 and 300 Hz, respectively. By sweeping the modulator up and down using makeLFO( ), you can hear these sidebands as they ascend and descend, sweeping across the rich audio spectrum of the incoming signal.

Frequency modulation (FM) is somewhat more complex than ring modulation, but its implementation is easy to understand. Discovered by John Chowning, FM synthesis involves three key components: A carrier frequency, modulator frequency, and modulation index. With ring modulation, we noted the presence of only one upper and one lower sideband, however with FM, there are many more sidebands present in the audio spectrum, producing richer timbral content. The incoming carrier frequency is modulated by the modulator at a given frequency, while the index calculates the range of deviation between the carrier frequency and the modulator frequency, or carrier/modulator. Think of the modulator as a rate of vibrato on the signal and the index as the depth of vibrato. Several factors go into changing the timbre of the resulting sound, such as the waveforms used as

the carrier, modulator, or the index number, but with a simple implementation comes a cornucopia of sounds. In fact, John Chowning licensed his patent for FM synthesis to Yamaha, who in turn created the powerful DX-7 synthesizer, among other instruments.

FM synthesis was highly desired, because you could obtain a rich variety of sounds — brassy, bell-like, percus-

```
rtsetparams(44100, 2)
load("FMINST")

duration = 2.0
amplitude = 1000
envelope = maketable("window", 1000, "hanning")
carrier = 10
modulator = 110
minindex = 0
maxindex = 1.0
waveform = maketable("wave", 1000, "sine")
index_envelope = maketable("line", "nonorm", 1000, 0,0, 0.5,1.0, 1.0,0)

increment = 0.5
for(start = 0; start < 20; start += increment){
     FMINST(start, duration, amplitude*envelope, carrier, modulator,
     minindex,maxindex, pan = random(), waveform, index_envelope)

     modulator += 110
     maxindex += 0.25
     }
```

Score file 48: Frequency modulation with subsonic carrier

It is interesting to note how in this score we are using a subsonic carrier frequency of 10 Hz, which still produces a dazzling array of sidebands given the sine wave modulator moving up the harmonic series at 110 Hz, with a low modulation index.

Here is one last FM scorefile, which establishes an interesting groove via a loop and the use of a "voltage-controlled" filter, designed after the famous Moog VCF.

Be careful of the resonance on this filter. One of the defining characteristics of MOOGVCF( ) is its sharp,

```
rtsetparams(44100, 2)
load("FMINST")
load("MOOGVCF")
load("PAN")
srand()

//-------------- FMINST
bus_config("FMINST", "aux 0-1 out")
start = 0
duration = 0.125
modulatorfrequency = 440
minindex = 0
waveform = maketable("wave", 1000, "sine")
guide = maketable("line", "nonorm", 8, 0,1.0, 1.0,0)
increment = duration
for(start = 0; start < 500; start += increment){
     carrier = cpspch(pickrand(6.02, 7.05, 5.07, 5.11, 5.02, 5.04, 5.09, 5.11))
     maxindex = irand(0,5)
     amplitude = trand(5000,20000)
     FMINST(start,   duration, amplitude, carrier, modulatorfrequency, minindex,
     maxindex, pan = 0, waveform, guide)
     FMINST(start+1, duration, amplitude, carrier, modulatorfrequency, minindex,
     maxindex, pan = 1, waveform, guide)
     }

//------------- MOOGVCF
bus_config("MOOGVCF", "aux 0-1 in", "aux 2-3 out")
start = 0
instart = 0
duration = 360
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, 0.25,1.0, 0.9,1.0, 1.0,0)
inputchannel = 0
pan = 0.5
```

```
bypass = 0
centerfrequency = makeLFO("tri", 0.125, 500,12000)

// careful with these values and keep below 1.0
resonance = maketable("line", 1000, 0,0.45, 0.25,0.1, 0.625,0.425, 1.0,0.25)

MOOGVCF(start, instart, duration, amplitude, inputchannel, pan, bypass,
centerfrequency,resonance)

//-------------  PAN
bus_config("PAN", "aux 2-3 in", "out 0-1")
start = 0
instart = 0
duration = 360
amplitude = 1.0
inputchannel = 0
pantype = 1
pan = maketable("random", 1000, "even", 0,1)
PAN(start, instart, duration, amplitude, inputchannel, pantype, pan)
```

Score file 49: FM groove with MOOGVCF( )

The world of synthesis and modulation is far reaching and ripe for more exploration. While the basic tenets of synthesis as outlined in this Interlude aren't themselves malleable, what you do with those tenets in your score files is what makes for interesting sonic results. We've just seen how frequency modulation can produce velvety, cascading waves or be used as the backdrop for a groove-based etude. Depending on the types of waveforms that you utilize —whether of our basic shapes or something constructed algorithmically or from known data plots — you'll find that your results will be surprising, interesting, and most importantly, fun.

Interlude V: Synthesis etude

Create a 2-3' etude using at least one of the synthesis or modulation procedures outlined in this section, as well as one that hasn't yet been covered. Scour the online documentation for your new instrument's implementation and be prepared to present its basic tenets and use for your peers.

|| Sonata VI: Randomness and algorithms ||

RTcmix's basic methodology lies in the calling of score file commands and instruments using the MINC parser with real-time handling of p-field parameters. The elements of C that are found within MINC — not to mention Python — afford us as users the ability to create well designed algorithms and processes that will enhance our sound world over time.

An algorithm is a process or order of operations that brings about a desired result. We've already encountered one example when we explored and wrote out the 3n + 1 wondrous number game in score file 16. In it, we not only observed a set of numbers that were derived from the algorithm itself, but also explored randomness by way of trand( ) for determining the original number that onsets the process. Algorithms are limited only by your imagination and in this Sonata, we will more deeply explore randomness and algorithmic procedures, creatively utilizing their principles in your work.

By way of review, we are already familiar with, or have come across srand( ), irand( ), trand( ), random( ), pickrand( ), as well as maketable("random") and some of its weighted distributions. I'd say that we're off to a pretty good start! However, let's begin exploring randomness in more detail by focusing on pickrand( ), which has been introduced but not fully explained.

> In reality, we're never really choosing at random since we're seeding our random number generator to something

Recall that an array in MINC is a series of elements that we can draw from in our score files. If, for example, I'd like to create a C major pentatonic melody, but choose randomly from the set of pitches that comprise the scale itself, I could begin by using pickrand( ).

Carefully copy the following code and run it a few times in your Terminal to compare the results.

```
srand()
print_off()

notes = {}
for(i = 0; i < 10; i += 1){
    note = pickrand(8.00, 8.02, 8.04, 8.07, 8.09)
    notes[i] = note
    }
print_on()
print(notes)
```

You'll see that each time you execute this score file, you don't have a great deal of control over the notes that are chosen at random as we fill our array with random pitches from pickrand( ). However, you can utilize another command called pickwrand( ), which chooses elements based on weighted distributions.

```
note = pickwrand(8.00,10, 8.02,10, 8.04,20, 8.07,20, 8.09,40)
```

Each note in our original set now has a corresponding number to go along with it, which you'll see I've written in pairs. 7.00 with a corresponding 10 literally means "choose 7.00 10% of the time." From this, you can see that I'm favoring the notes E4, G4, and A4 a bit more than C4 or D4, and A4 will be picked

about 40% of the time. Go ahead and substitute this new line into score file 50 and check the results.

While it is exciting to know that we can more carefully craft our parameters using random numbers, let's go one level deeper and create some of these distributions from scratch.

> The possibilities are now starting to take shape as we consider these random distributions. What if, for exam-

We'll start by choosing random integers between 0 and 10 using trand( ) and aim to favor a return of values in the middle, a so-called triangular distribution due to its high probability of middle values and low probability of low and high values. When the probability distribution is viewed on a graph, it resembles the shape of a triangle, hence the name.

```
srand()
print_off()

gamut = {}
for(i = 0; i < 10; i += 1){
    x = trand(0,11)
    y = trand(0,11)
    element = ((x + y) / 2)
    gamut[i] = element
    }

print_on()
print(gamut)
```

> I added the print_off( ) and print_on( ) commands to ensure that all that is printed to the Terminal is a

In order to get at values weighted toward the middle, we first choose two random integers and then take the average of them. Since we're finding a generic list of elements, I've gone with the variable name gamut to denote our array, filling it with the element variable. Careful readout of your Terminal will show that be-

cause we're taking the average of two integers, we'll at times end up with floating point numbers, but we can mitigate this by rounding up the values for each individual element.

```
element = trunc((((x + y) + 0.5) / 2))
```

It's been a while since we've seen trunc( ), but recall that it takes off decimal points, which is our goal here, only after we've rounded up by adding 0.5 to each of our elements.

We can also weight outcomes toward the low or high values of our list.

```
srand()
print_off()

gamut = {}
for(i = 0; i < 10; i += 1){
    x = trand(0,11)
    y = trand(0,11)

    if(x > y){
        element = x
        }

    else{
        element = y
        }

    gamut[i] = element
    }
```

```
print_on()
print(gamut)
```

We're again using two random integers, this time ensuring that each number for element will be drawn from the highest value returned between x and y, rather than their average. This is accomplished by our conditional statement "if x is greater than y, then element equals x, else element equals y." Conversely, we can change the outcome for element to favor the lower end of the 0-10 spectrum, but I'll pause here and let you come up with the solution to that on your own.

The preceding examples will generate one single outcome, which we can of course transfer to a loop and then an array to garner a series of elements. However, we do have the ability to generate and manipulate random numbers in the maketable( ) command.

```
gamut = maketable("random", 10, "gaussian", 0,1)
dumptable(gamut)
```

A Gaussian distribution is sometimes referred to as a normal distribution and is one of the tried and true "bell curves" that you might be familiar with w/r/t grades. Like triangular distributions, it has a proclivity toward values in the middle of a given range, albeit with slightly different outcomes. The dumptable( ) command will print the contents of your table to the Terminal window. Moreover, there are a few commands that will allow you to further manipulate your table data, much like what we saw earlier with makefilter( ).

> Returning individual Gaussian numbers exists in the JFUNCS( ) library.

```
gamut = maketable(“random”, 10, “cauchy”, 30,100)
gamut_plus_one = add(gamut, 1)
gamut_minus_two = sub(gamut, 2)
gamut_times_five = mul(gamut, 5)
gamut_dividedby_two = div(gamut, 2)

dumptable(gamut)
dumptable(gamut_plus_one)
dumptable(gamut_minus_two)
dumptable(gamut_times_five)
dumptable(gamut_dividedby_two)
```

Note that we've selected a range of 30 to 100 using a Cauchy distribution, which also has a bell-curve shape, and in this example, we can interpret the outcomes for any of a variety of musical parameters, including frequency in Hz. If we want to utilize those values as pitch data for an instrument, we can convert it accordingly.

```
pitch_gamut = makeconverter(gamut, “cpspch”)
```

This will translate all of our values in Hz to octave point pitch class, so that, for example, a value of 440 will return 8.09. A check of the online documentation will show that there are a number of ways to convert your table data to fit your particular needs.

If an algorithm is a process or order of operations that will produce an outcome, then it is important to consider that a well-considered and strongly crafted algorithm will lead to more effective results. For example, we might implement a random walk to determine pitch, as we did in score file 15.

```
srand()
```

```
print_off()
pitches = {}
note = 60 //middle C
for(i = 0; i < 24; i += 1){
    x = random()
    if(x < 0.5){
        note += 1  //50% of the time, increase by a half step
        }

     if(x > 0.5){
            note -= 1  //50% of the time, decrease by a half step
        }

    pitches[i] = note
    }

print_on()
print(pitches)
```

This particular random walk will return notes one half step up or one half step down fifty percent of the time, respectively. This is akin to a coin flip, which will vary over time, but we can reasonably expect our algorithm to have 12 ascending moments and 12 descending through the loop. We've laid a backdrop for our process, but this is ripe for further refinement. What if, for example, you wanted to favor, ever so slightly, ascending notes rather than descending? What ratio will you choose? 51/49? 60/40? Could all ascending notes move up by a whole step while those that descend move by a half step?

Moreover, let's say that we're writing this wonderful melody for an instrument that has a limited range. We'll go with an octave, so we can only choose notes between 60 and 72. We're going to need to set up some boundaries and another rule for our random walk once it reaches a boundary point. There are any of a number of ways to treat this rule, but we'll go with returning our note to

F#, in the middle of our octave range and a point where we can start this new melody.

```
srand()
print_off()
pitches = {}
note = 66 //F# above middle C
for(i = 0; i < 24; i += 1){
    x = random()
    if(x < 0.54){
        note += 2  //50% of the time, increase by a half step
        }

     if(x >= 0.46){
        note -= 1  //50% of the time, decrease by a half step
        }

    if(note > 72 || note < 60){
        note = 66
        }
    pitches[i] = note
    }

print_on()
print(pitches)
```

We can continue refining and transforming this random walk as much as we'd like. In order to avoid getting caught up in the myriad of possibilities, I find it helpful to keep a composition journal that I write in plain english before crafting on the computer. For this example, I might write something like, "I want a limited range of MIDI pitches between 60 and 72 that will revolve around 66, or F#. Using a random walk that slightly favors ascending pitches in whole steps — as opposed to descending half steps — I'll create some boundary lines that force the melody back to F# if it surpasses the octave range. I'm thinking about tying

durations into this as well, so that each whole step corresponds with a longer note duration and greater amplitude each time I reach F#. I'll try this out on a MMODALBAR( ) instrument and maybe add REVERBIT( )..." You get the idea.

The study of chaos theory is incredibly interesting. Using some of its precepts, we are able to reasonably predict dynamic outcomes that are highly sensitive to initializations. One such algorithm that falls under the umbrella of chaos theory is the so-called logistic map, which among other uses can predict population density based on initial conditions. While the scope of this text isn't of course to take any of these sociological issues into consideration, we do find the the logistic map is incredibly easy to implement into MINC and provides some wicked fun results from minor deviations.

Here is the equation itself, which is actually quite simple.

$$xt + 1 = Rxt(1 - xt)$$

We start with initializing two variables, R, which represents a number that corresponds with the combined effects of birth/death rates, and xt, fun concept called the "fraction of carrying capacity," which is related to the current population size.

In MINC, we'll use a loop to get our successive values for x and fill them in an array.

I'm culling these variables, their explanations, and the sample values from a fantastic book by Melanie Mitchell

```
R = 2.0
x = 0.5

gamut = {}
for(i = 0; i < 20; i += 1){
    x = ((R * x) * (1 - x))
```

```
    gamut[i] = x
    }
print(gamut)
```

Our Terminal readout provides nothing to write home about at all, as you'll note a sequences of 0.5 values.

```
[0.5, 0.5, 0.5, 0.5, 0.5, 0.5, 0.5, 0.5, 0.5, 0.5]
```

This is just a starting point for the fun involved with this equation, and speaks, algorithmically, to the beauty of chaotic systems and how they deviate based on minute changes in initial variables. Try, for example, changing x to 0.2 and check out the results.

```
[0.32, 0.4352, 0.49160192, 0.499858944505, 0.499999960207, 0.5, 0.5, 0.5, 0.5, 0.5]
```

It's interesting how we start below 0.5 and then slowly but surely make our way toward it before the string of repetition onsets. As noted by Melanie Mitchell, if we let R = 3.1 and x = 0.2, we find that the returning values will eventually oscillate around 0.55 and 0.76.

```
[0.496, 0.7749504, 0.540647060374, 0.769878231097, 0.549213795177, 0.767491807329,
0.553189212337, 0.766229813842, 0.555277227287, 0.765527727245]
```

With values for R up to 3.4, this oscillation phenomena will continue. When R is between 3.4 and 3.5, x will oscillate between four values. With R between 3.54 and 3.55, x will oscillate between eight values, between 3.564 and 3.565, six-

teen values. Thus, x is relatively predictable for these values for R, but chaos ensures once R = 3.569946, when x no longer hovers between a set of values and thus shows more chaotic behavior. While at first blush these values might seem to be completely random, they in fact aren't, but closely resemble random behavior.

```
R = 3.569946
x = 0.5

gamut = {}
for(i = 0; i < 20; i += 1){
    x = ((R * x) * (1 - x))
    gamut[i] = x
    }
print(gamut)
```

```
[0.8924865, 0.34255183839, 0.803987811424, 0.56259282557, 0.878499944893,
0.381048152586, 0.841973399124, 0.474996288695, 0.890254621243,
0.348788474365, 0.810859950725, 0.547508663226, 0.884428870981,
0.364900042489, 0.827327850893, 0.509989912316, 0.892130225286,
0.343549777878, 0.805106102693, 0.56016117682]
```

What an equation! Let's take our gamut of data and convert it into something musically significant, such as pitches for the sitar physical model instrument, MSITAR( ) . Before we do that, however, we'll first need to get our individual pitches into a range that will reasonably fit into MIDI notes and then Hz. An example might look something like this:

```
pitches = gamut * 100
current_pitch = cpsmidi(trunc((pitches[ii]) + 0.5))
```

Now we can combine our processes — the logistic map, conversion to a fitting pitch range — into an example score file that includes all of the requisite p-field data for MSITAR( ).

```
rtsetparams(44100, 2)
load("MSITAR")

R = 3.569946
x = 0.5

pitches = {}
number_of_pitches = 20

//parameters for MSITAR
duration = 5.0
amplitude = 20000
envelope = maketable("line", 100, 0.0,0.125, 0.75,0.75, 1.0,0.0)
pluck = 0.25

for(start = 0; start < number_of_pitches; start += 1){
     x = ((R * x) * (1 - x))
     pitches[start] = x * 100
     current_pitch = cpsmidi(trunc((pitches[start]) + 0.5))
     MSITAR(start, duration, amplitude*envelope, current_pitch, pluck)
     }
```

Score file 50: MSITAR( ) pitches from logistic map

If that isn't the apt soundtrack for a nightmare sequence in a horror film, I don't know what is.

Without a doubt, random parameters and algorithmic procedures play a large role in much of the music being composed today. With an abiding sense of curiosity and willingness to research and apply these processes in your own work, I hope that you will find as much joy and wonderment as I do while composing.

There is no doubt that leaving some musical parameters and elements to chance greatly enhances our experience with our music and yields surprising, if ultimately fitting, results.

Sonata VI: Algorithmic etude

Compose a one to three minute work that includes one, if not two algorithmic processes. You might spend some time searching for an interesting equation or process that can be implemented in MINC and applied to at least two different RTcmix instruments. Moreover, utilize at least three different random processes or weighted random distributions in your finished composition.

|| Interlude VI: Pitch transformations ||

Interlude III left us with a complex Python script that outlined a series of pitch transformations using the twelve tone system. In the early twentieth century, Viennese composer Arnold Schoenberg codified a new method of composition that would finally divorce Western classical music from its evolution of tonality and fully liberate chromaticism, which is referred to as twelve tone music, serialism, or dodecaphony.

No worries if that all sounds super confusing, as we'll take time to fully explain the notion of pitch and some of the meaningful ways — including Schoenberg's method — that we can manipulate pitches in our music using RTcmix.

Let's start by rehashing the beginning of Sonata III, where we introduced the major scale. Recall that when looking at a piano, we see black and white keys, with the black keys laid out in alternating groups of two and three. Looking at the group of two black keys, we know that the white key adjacent to the leftmost black key is C. Playing from that particular key up to the next C produces our requisite sequence of half steps and whole steps that constitutes a major scale with the letter names C, D, E, F, G, A, B, and again, C representing each individual pitch.

The low C to the higher C spans an octave and if we include all of the black and white keys in that octave, we have a total of eleven pitches, or twelve if you count the C one octave higher than where we started.

Play them all in succession and you've just played a chromatic scale!

If we put each of the notes with their letter name into an array, we might get something that looks like this:

```
notes = {“C4”, “C#4”, “D4”, “D#4”, “E4”, “F4”, “F#4”, “G4”, “G#4”, “A4”, “A#4”, “B4”}
```

However, let's abstract our notion of pitch by divorcing each from its letter name and instead give each note in the chromatic scale a numerical designation.

```
notes = {0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11}
```

Now we can see that each C will equal 0, each F will equal 5, each B will equal 11, and so on. This is an important step, as we want to get used to seeing 6 and immediately thinking F#, or knowing that Bb equals 10, and we're not super concerned with octave designation right now, since we want to be able to instead think of these notes as abstract entities that fit within one octave.

> For those who don't have a background in music theory, the # and b can enharmonically represent the

There are many different ways that we can manipulate this array or list of pitches. For example, we can divide our pitches into subsets of the full gamut, creating chords. For example, a C major chord could be constructed out of our array like this:

```
notes = {0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11}
c_major_chord = {notes[0], notes[4], notes[7]}
```

Or, we might randomly select three notes from our array to make our chord.

```
srand()
chord = {}
notes = {0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11}
array_length = len(notes)

for(i = 0; i < 3; i += 1){
    new_note = notes[trand(0,array_length)]
    chord[i] = new_note
    }
print(chord)
```

Our code also works well in Python.

```
from rtcmix import *
import random
srand()
chord = []
notes = [0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11]
list_length = len(notes) #returns 12, so take away one to get indices 0-11
print(list_length)

for i in range(0,3):
    new_note = notes[random.randint(0,list_length - 1)]
    chord.append(new_note)

print(chord)
```

If we'd rather avoid repeated notes, we can always call upon spray_init( ), but this will work well for now.

One effective composition method of manipulating pitches is to play your particular succession in reverse.

```
pitch_gamut = {0, 1, 2, 3, 4, 5}
array_length = len(pitch_gamut)
gamut_reversed = {}
for(i = 0; i < array_length; i += 1){
    gamut_reversed[i] = pitch_gamut[((array_length - 1) - i)]
    }

print(pitch_gamut)
print(gamut_reversed)
```

And in Python:

```
from rtcmix import *
pitch_gamut = [0, 1, 2, 3, 4, 5]

def reverse(pitches):
    return pitches[::-1]

gamut_reversed = reverse(pitch_gamut)

print(pitch_gamut)
print(gamut_reversed)
```

Let's take a pause to focus on Python for a second. See how thrifty the Python code looks? This is because we were able to create a function that reverses lists — thus precluding our need for a loop — using a handy device called list slicing. All arrays in MINC can be indexed, but in Python, we can uniquely index our lists both forwards and backwards. So for example if we wanted to access the first element in our gamut of pitches — just like in MINC — we'd use something like this:

```
pitch_gamut = [60, 64, 67, 68, 69, 70]
first_pitch = pitch_gamut[0]
```

We can can access “68” in two ways, either by our familiar method of reading from left to right starting at 0, or now backwards.

```
pitch = pitch_gamut[3]  //index 3 = 68
pitch = pitch_gamut[-3] //returns 68 as well
```

What if we wanted pitches 64, 67, and 68 in our gamut? For this, we will “slice” the list.

```
chord = pitch_gamut[1:4]
```

Note how we’re gathering elements one through three in the list. It’s a bit of a misnomer, since the list slice looks like it’s saying “take elements one through four,” so you might at first blush expect to return [64, 67, 68, 69]. Instead it’s best to think that we’re starting with pitch_gamut[x] and ending with pitch_gamut[x - 1], which can be confusing if you’ve been rocking out on MINC up until now.

To access our list verbatim, we can simply write:

```
repeat_gamut = pitch_gamut[:]
```

If you leave one end of the slice empty, Python will return only the first or last set of elements that you are looking for, respectively.

```
first_three_pitches = pitch_gamut[:3]
last_three_pitches = pitch_gamut[3:]
last_two_pitches = pitch_gamut[4:]
```

We'll return to some heavier Python when the time comes, but let's get back to pitch transformations in general. We've looked at denoting subsets of our gamut of pitches and now utilizing them in reverse, but we can also invert them.

Inversion of pitches is a simple notion that merely involves addition and subtraction, but with a caveat called mod-12 arithmetic. When we describe pitches in western music, you'll find that we only represent them with the letters A, B, C, D, E, F, and G. Rather than call the next note in this succession "H," we instead wrap around to "A" again and repeat our succession. Similarly, in mod-12, we wrap any results above 12 back into the range of 0 - 11, which of course corresponds nicely to our notion of chromaticism. For example, $8 + 5 = 13$ in our numbering system, but in mod-12, $8 + 5 = 1$. To understand this, you can think of reaching 12 and then wrapping your way back down to 1, just like we do on a clock face. Similarly, $8 + 4 = 0$ in mod-12.

Now we're ready to invert our pitches and to do so we'll need the mod( ) function in MINC and the math.fmod( ) function in Python.

```
gamut = {0, 5, 4, 9, 11}
gamut_length = len(gamut)
invert_gamut = {}
for(i = 0; i < gamut_length; i += 1){
    invert_gamut[i] =  mod(((gamut[i] - 12) * -1), 12)
    }

print(gamut)
print(invert_gamut)
```

And again in Python.

```
from rtcmix import *
import math

pitch_gamut = [0, 5, 4, 8, 2]

def invert(pitches):
    return [math.fmod(((x - 12) * -1), 12) for x in pitches]

print(pitch_gamut)
print(invert(pitch_gamut))
```

Transposing pitches is another useful method for playing around with pitch and is easy to implement now that we have mod-12 in our tool belt.

```
pitch_gamut = {0, 5, 4, 8, 2}
gamut_length = len(pitch_gamut)
transpose_gamut = {}
transposition = 2  //transpose by a whole step, or Major 2nd

for(i = 0; i < gamut_length; i += 1){
    transpose_gamut[i] = mod(pitch_gamut[i] + transposition, 12)
    }

print(pitch_gamut)
print(transpose_gamut)
```

Now for Python.

```
from rtcmix import *
import math

pitch_gamut = [0, 5, 4, 8, 2]
transpose_gamut = []

transposition = 7 # up a Perfect 5th
```

```
def transpose(input_list, transposition_value):
    return [math.fmod(element+transposition_value, 12) for element in input_list]

print(pitch_gamut)
print(transpose(pitch_gamut, transposition))
```

Before we consider Schoenberg's method of serialism, let's review. We can declare arrays or lists of pitches and have the ability to transpose them by a given value. We can also write them backwards, or in reverse, and can invert them. As we now consider dodecaphony, we'll codify these transformations using some specific methods and place each of our lists or arrays of pitches into a suitable range for RTcmix to play back as MIDI notes.

For Schoenberg, composition with twelve tones eliminated the need to write music that centered around a central, tonal key area. Instead, his method relies solely on each of the pitches of the chromatic scale and one is not permitted to repeat a pitch until each note in the chromatic collection has been utilized. Thus, you can see how his mode of operation totally and completely liberates chromaticism as the central focus of the music.

We begin with a twelve note row, sometimes referred to as an aggregate, which we've up until now been calling array, list, or gamut. Just as we encountered with algorithms, a well-crafted, carefully considered aggregate row form will assist greatly in the outcome of your work. So, let's throw caution to the wind and make a row using chance procedures!

```
srand()
spray_table = 1
spray_size = 12
seed = trand(0,100)
spray_init(spray_table, spray_size, seed)
```

```
aggregate = {}

for(i = 0; i < spray_size; i += 1){
   aggregate[i] = get_spray(spray_table)
   }

p_0 = aggregate
```

Here is my aggregate row form, which is also called prime form. It's customary to say that an non-transposed, prime row is called p0, hence our variable p_0.

Next, we'll take our familiar code from above for writing our notes backward and utilize it to get the retrograde row form, or r0 if it's not transposed.

```
retrograde = {}
for(i = 0; i < spray_size; i += 1){
    retrograde[i] = aggregate[((spray_size - 1) - i)]
    }

r_0 = retrograde
```

From there, we can can also invert our pitches to get the inverted row form, or i0.

```
inversion = {}
for(i = 0; i < spray_size; i += 1){
    inversion[i] =  mod(((aggregate[i] - 12) * -1), 12)
    }

i_0 = inversion
```

The notion of retrograde inversion row form is the last component of the twelve tone method. We'll take the inverted row form and write it backward, designated as ri0.

```
retrograde_inversion = {}
for(i = 0; i < spray_size; i += 1){
    retrograde_inversion[i] = inversion[((spray_size - 1) - i)]
    }

ri_0 = retrograde_inversion
```

So far so good! Here's the entire score file tidied up with some print statements to verify our work.

```
srand()
spray_table = 1
spray_size = 12
seed = trand(0,100)
spray_init(spray_table, spray_size, seed)

aggregate = {}

for(i = 0; i < spray_size; i += 1){
   aggregate[i] = get_spray(spray_table)
   }

p_0 = aggregate

retrograde = {}
for(i = 0; i < spray_size; i += 1){
    retrograde[i] = aggregate[((spray_size - 1) - i)]
    }

r_0 = retrograde

inversion = {}
```

```
for(i = 0; i < spray_size; i += 1){
    inversion[i] =  mod(((aggregate[i] - 12) * -1), 12)
    }

i_0 = inversion

retrograde_inversion = {}
for(i = 0; i < spray_size; i += 1){
    retrograde_inversion[i] = inversion[((spray_size - 1) - i)]
    }

ri_0 = retrograde_inversion

print(p_0)
print(r_0)
print(i_0)
print(ri_0)
```

Score file 51: Twelve tone transformations in MINC

Our final task is to put these into a suitable range for MIDI. In Python, we can accomplish this by creating a new function for MIDI conversion, as we did in score file 21. I like to think of these as starting in octaves spanning C to C, so that I'm transposing into middle Cs range (60) or an octave higher (72) or lower (48). In MINC, you can simply add each requisite transposition into your for( ) loop while constructing your row forms.

```
rtsetparams(44100, 2)
load("MMODALBAR")

//---------- MMODALBAR
start = 0
duration = 1
amplitude = 30000
```

```
//---pitch stuff
spray_table = 1
spray_size = 12
seed = 12

aggregate = {}
transposition = 48
spray_init(spray_table, spray_size, seed)
for(i = 0; i < spray_size; i += 1){
   aggregate[i] = get_spray(spray_table) + transposition
   }

reverse = {}
invert = {}
retrograde_invert = {}
for(i = 0; i < spray_size; i += 1){
    reverse[i] = aggregate[i - 1] + 60
    invert[i] =  aggregate[i - 12] + 72
    }

retrograde_invert = {}
for(i = 0; i < spray_size; i += 1){
    retrograde_invert[i] = invert[i - 1] + 60
    }

hardness = 1.0
position = 1.0
instrument = 4
pan = 0.5

for(start = 0; start < spray_size; start += 1){
    MMODALBAR(start, duration, amplitude, aggregate[start], hardness, position,
    instrument, pan)
    MMODALBAR(start, duration, amplitude, reverse[start], hardness, position,
    instrument, pan)
    MMODALBAR(start, duration, amplitude, invert[start], hardness, position,
    instrument, pan)
    MMODALBAR(start, duration, amplitude, retrograde_invert[start], hardness, position,
    instrument, pan)
    }
```

Score file 52: Score file 21 in MINC

Neither score files 21 or 52 might be the most musically edifying composition right now — just a roving series of MMODALBAR( ) chords built from dodecaphonic transformations — but you can now imagine all of the possibilities that you have at your disposal for further refinement. You could certainly transpose any of these row forms, change durations, amplitudes, instruments, add effects, etc.

> If you transpose your prime form up two half steps, you are customarily creating p2. Similarly, if you trans-

We'll sum up this section with a return to Python and some of the incredibly useful pitch manipulating methods that you can implement thanks to Python's ability to define functions. The following examples were culled from Christopher Burns' "Compositional Algorithms" course at the University of Wisconsin-Milwaukee, which I had the pleasure of taking in 2007. The course was designed around the LISP programming language, which I've adapted to Python for use with RTcmix.

We can place conditional tests within our functions to return values based on their outcomes. For example, if I wanted to return the absolute value of a given number, I could create a simple test to see if the number in question is either positive or negative, knowing that tests for absolute value return the number's distance from zero, which will intrinsically be positive.

```
def absolute_value(x):
    if(x >= 0):
        return x
    else:
        return x * -1
```

```
print absolute_value(-5)
```

This is useful because we can now generate interesting lists of numbers that we can turn into any form of musical data that we wish, such as generating lists of numbers according to the Fibonacci sequence up to a determined location.

```
fibonacci_list = []
def fibonacci(x):
    if(x == 0):
        return 0
    elif(x == 1):
        return 1
    else:
        return (fibonacci(x - 1) + fibonacci(x - 2))

for i in range(0,10):
    fibonacci_list.append(fibonacci(i))

print fibonacci_list #first ten elements in the fibonacci sequence
```

From here, we can return to list slicing to hone in an any portions of the fibonacci sequence that we'd like to use in our score. In this instance, we'll can take our list of fibonacci numbers and return them in a palindrome.

> For example, take a moment to figure out how you would isolate the numbers 21, 34, 55, and 89 in the list of

```
def palindrome(input_list):
    return input_list + input_list[::-1]

print palindrome(fibonacci_list)
```

However, note that we've repeated the top number in our list, 34, as we appended the original list with its reversed form. To mitigate this dilemma, we can elide the repeated note all together.

```
def palindrome_elided(input_list):
    first_half = input_list
    reverse_list = input_list[::-1]
    second_half = reverse_list[1:]
    return first_half + second_half

print palindrome_elided(fibonacci_list)
```

We've failed to thus far utilize repetition of our list of numbers, useful for our minimalists at heart.

```
def repeat(input_list, iterations):
    return input_list * iterations

print repeat(fibonacci_list, 4)
```

The following score file example sums up our newly codified transformations using an aggregate twelve note row while also defining functions that will divide the row into hexachords and rotate the pitches in the hexachords to the left or to the right.

```
from rtcmix import *

sample_series = (0, 11, 6, 2, 9, 5, 10, 4, 7, 3, 1, 8)

def reverse(input_list):
```

```
    return input_list[::-1]
print reverse(sample_series)

def get_first_hexachord(input_series):
    return input_series[:6]
print get_first_hexachord(sample_series)

def get_second_hexachord(input_series):
    return input_series[6:]
print get_second_hexachord(sample_series)

def get_cheap_second_hexachord(input_series):
    return reverse(get_first_hexachord(input_series))
print get_cheap_second_hexachord(sample_series)

def transpose(input_list, transposition_value):
    return [(element + transposition_value) % 12 for element in input_list]
print transpose(get_first_hexachord(sample_series), 10)

def rotate_right(input_list, rotation):
    return input_list[rotation:] + input_list[:rotation]

print get_second_hexachord(sample_series)
print rotate_right(get_second_hexachord(sample_series), 1)

def rotate_left(input_list, rotation):
    return input_list[-rotation:] + input_list[:-rotation]

print get_second_hexachord(sample_series)
print rotate_left(get_second_hexachord(sample_series), 2)
```

Score file 53: Complex transformations in Python

Interlude VI: Pitch etude

Compose a brief, one minute RTcmix etude that focuses on pitch and various transformations. Use two instruments that are unfamiliar to you or that you have yet to use by searching the online documentation. Think of your etude as a small invention for two voices, so that the pitches between your two instruments interact in a meaningful, contrapuntal fashion.

|| Sonata VII: RTcmix in Pure Data and Max ||

Visual programming languages afford the ability to create complex patches — akin to our score files or scripts — using a set of patchable objects, messages, and other tools designed for real-time control and signal processing. Pure Data (Pd) and Max/MSP (Max) are incredibly popular platforms for realizing your work using the visual programming paradigm and RTcmix has been adapted for use within either of these powerful software packages.

In the mid 1980s, Miller Puckette was at IRCAM in Paris, working on developing a suite of patchable tools for realizing live, interactive computer music which was eventually dubbed Max — in homage to Max Mathews — and has been further refined by David Zicarelli and made available, commercially, through his company Cycling '74. Once Max was able to realize the handling of digital signals, it was then called Max/MSP and after adding libraries of video processing tools, has been known as Max/MSP/Jitter, though "Max" or "Max/MSP" are interchangeable titles and in the computer music world, saying that you work with Max is very common and most anyone will know what you are referring to.

In 1996, Miller redesigned his code and began offering the open source program Pure Data, which he continues to maintain under the heading "Pd Vanilla" due to its set functionality, which differentiates it from "Pd-extended," a community-curated version that offers a large set of external libraries, including the powerful GEM library for video processing. Whether you are working in Pd

Vanilla or Pd-extended, most will know what you are using if you indicate that you are working with Pd.

Despite their shared lineage, these two programs should really be thought of as separate entities. Due to its open source nature, Pd is — like RTcmix — completely customizable, and there is no shortage of documentation for learning how to write your own objects, add them to your library, or change the existing code that constitutes the program to personalize your own experience while using it. On the other hand, because Max/MSP is commercially available, it is curated by a team of paid developers, and since the recent release of Max 7, boasts a powerful GUI interface that is predicated on user accessibility — especially geared toward beginning programmers — and a wide range of pre-designed patchable modules for quickly realizing work in signal processing. Please know that whatever flavor you choose between Pd and Max is of course up to you.

Personally, I enjoy working in both Pd and Max. I first learned visual programming on Pd and continue to enjoy relying on it nearly exclusively for my work in live, interactive performances. Because RTcmix was first developed for Max and has only recently realized functionality in Pd, there was a period of time where I focused my work on Max if I was going to utilize RTcmix in conjunction with playing live. Again, there is no shortage of tools available to the computer musician who is working today and after sampling freely from both Pd and Max in this Sonata, my hope is that you'll find a platform that works best for you, aids in your enjoyment while programming and patching your music, and assists in realizing your tasks quickly and efficiently.

We'll begin by downloading both programs. Because Pd is free and open source, you'll be able to enjoy it cost free for as long as you'd like. Max will offer a trial version that has full functionality. While you will eventually need to pay for Max, there are deep discounts for students and educators alike.

Head to http://www.puredata.info and make sure to download either Pure Data, which is sometimes referred to as Pd-Vanilla. Now, do the same for getting Max by navigating to https://cycling74.com and downloading Max 7. Both download processes will be straightforward, in that you'll be getting a .dmg file that will take you step by step through the download process, much like you'll do for other software that you might get online.

We'll also need the RTcmix object that runs for each program, respectively. We'll first get the version for Max by going back to the RTcmix web page and clicking on the "rtcmix~" link. The heading for Mac OS X Directories will point to a download link for the requisite .zip file. Don't worry if that link says "max6" somewhere in its name, as it will run just fine in our Max 7 application.

When you downloaded Max, you might've missed the fact that you now have a folder in your Documents called Max 7. Within that folder are a few subfolders, one of which is called "Library." This is where you'll need to place the unzipped "rtcmix~_1.92/osx/max6" folder (be sure to unzip it by double-clicking on it) that we just downloaded and if you can't find it, chances are it is in your Downloads folder. The *raison d'être* of this Max 7 "Library" folder is to house all of your so-called external objects and libraries so that Max can find them once we call upon them in our work.

Now back to Pd. Navigate to https://github.com/jwmatthys/rtcmix-in-pd and click on the link in the lower right corner of your screen which denotes "Download ZIP." You'll be downloading a folder called "rtcmix-in-pd-master" which will for now reside in your Downloads. Create a folder in your Documents called "Pd-externals" and then move the "rtcmix-in-pd-master" folder into it.

Go ahead and open Pure Data and then open a new patcher window by clicking command-n or head to the Menu bar and clicking File → New.

Before we can have fun with RTcmix in Pd, we have one last step. Pd needs to be able to find your "Pd-externals" folder, so click Pd-extended → Preferences → Path in the Menu bar and create a new path for Pd that points to the aforementioned externals folder, then click "Apply" and "OK" and you should be good to go.

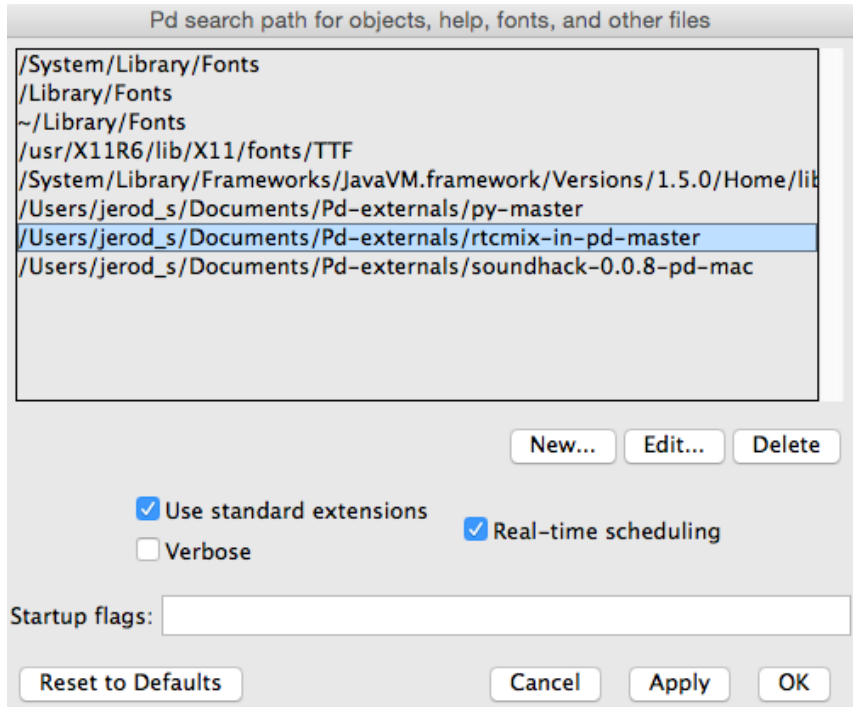

Figure 9: Pure Data "Path" setup

When you open Pd, you have two windows, one is the main Pd window, which depending on a variety of factors might already have some text printed to it, and the patcher window, where we will create and interact with our Pd patches. The Pd window exists to communicate with you if, for example, there is an error in your patch, but you may also print messages to it in order to verify that portions of your Pd patch are working correctly. I prefer collapsing the main Pd window a bit and maximizing the space I have on my Pd window for patching and we'll begin by creating our first object in the patcher window.

> There are way too many incredibly helpful tutorials, manuals, and YouTube videos dedicated to starting

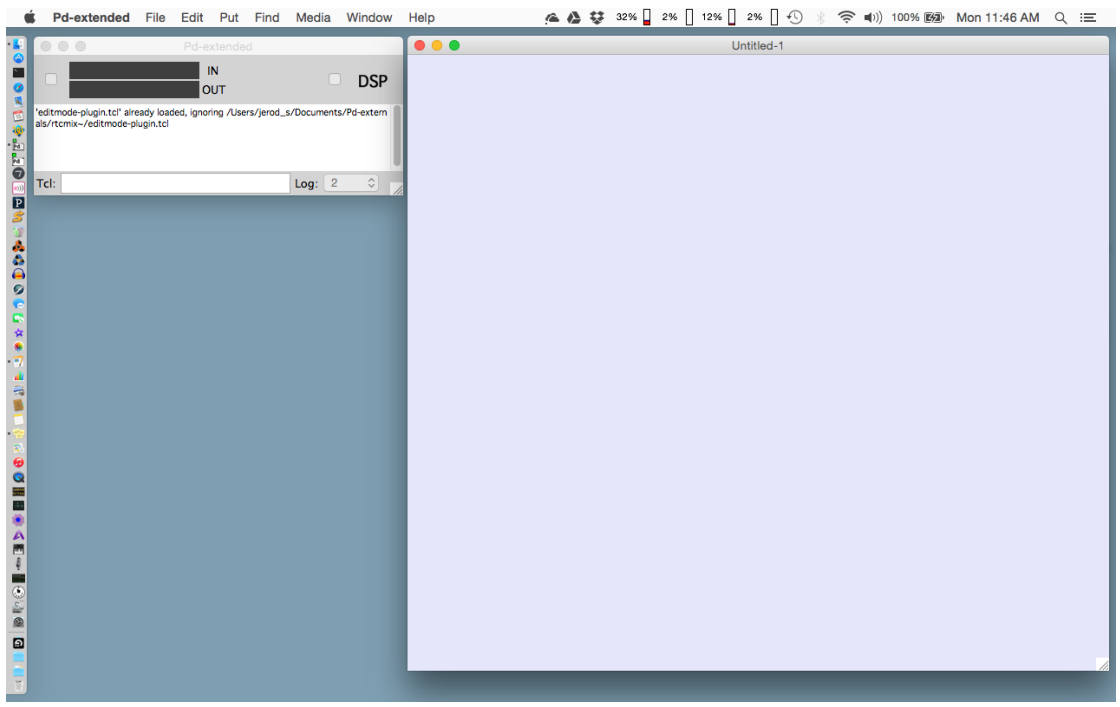

Figure 10: Pure Data workspace

Type command-1 and you'll notice that your mouse arrow has changed to a pointing hand with a rectangular box. This means that your are in edit mode and can create Pd objects, messages, number boxes, symbols, or comments. Pd runs in two modes, edit and run mode, which allows you to interact with your Pd patch once you've finished editing it. Type in "rtcmix~ 2" and click away from the rectangular box to create the object. Be sure to include a space between the tilde and 2.

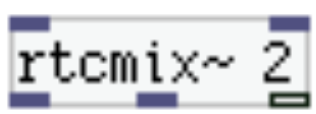

Figure 11: Pure Data object

For some of you, this is your first Pd object! There are a few things to keep in mind with Pd (and Max) objects. The first is that they contain inlets and out-

lets. Inlets can receive a variety of things, such as audio signals or messages, and outlets will output the result of the processes that implemented within the Pd object itself. This is part of the beauty of a visual programming language: We won't be confronted with lines of text as we've been getting used to in MINC and Python, but we're now engaging with that code as it is encapsulated within a tidy, graphical object. In addition, many Pd objects have a namespace (like rtcmix~ or metro or + or moses) and an argument, which we've denoted as 2 for our object. We didi this because we are specifying two so-called signal outlets that you can think of as functioning akin to the number of outlets we specify in rtsetparams( ).

Practice getting between edit mode and run mode by typing command-e. As you move the cursor around, you'll notice that edit mode features our little hand icon and run mode retains the familiar arrow. In edit mode, type command-2 to create a message and type in "bang." Click away from the message (which isn't rectangular in shape but looks like a tiny flag) to create it and hover over its outlet. You should notice a little target circle appear. Click and hold, then drag your cursor to create a patch line that can then be connected to the leftmost inlet of the [rtcmix~] object.

Figure 12: "bang" message connected to the RTcmix object

Create another message (command-2) that states "flush" and connect that to the leftmost inlet of [rtcmix~] as well. Lastly, create a [dac~] object (command-1) and connect the two signal outlets of the [rtcmix~] object to its corresponding inlets. The [dac~] object stands for digital-to-analog converter and translates the

digital stream of data coming from our computer to acoustic energy that we can hear via our speakers or headphones.

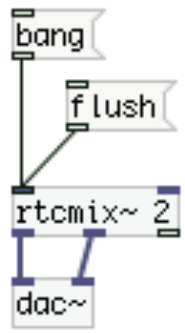

Figure 13: Adding a digital-to-analog converter

Perfect. Now we are able to go to run mode (command-e again if you aren't already there) and click on our [rtcmix~] object. A new editing window will pop up where you'll write all of your MINC code.

Feel free to copy and paste a completed score file here or start from scratch and build a simple oscillator like I've done. It's important to remember two things at this point. First, you no longer need to write rtsetparams( ) or use load( ) because that functionality is built into the Pd object itself (same goes for Max, of course). Also, we need to close this window after we're done editing to actually load the MINC code into the object itself.

Next, go to your main Pd window and click on the box next to "DSP" so that we can begin processing audio. We are ready to hear our score file once we click on the "bang" message that we created earlier. If all goes well, you should hear your RTcmix script running in Pd!

> And such a let down, if not. The likely culprit is found in the top Menu bar under Media → Audio Set-

You can either let the script run its course or click on "flush" to stop rendering it. Now that we have sound working in Pd and have heard our first script, right-click on the [rtcmix~] object and select "Help" from the list of options.

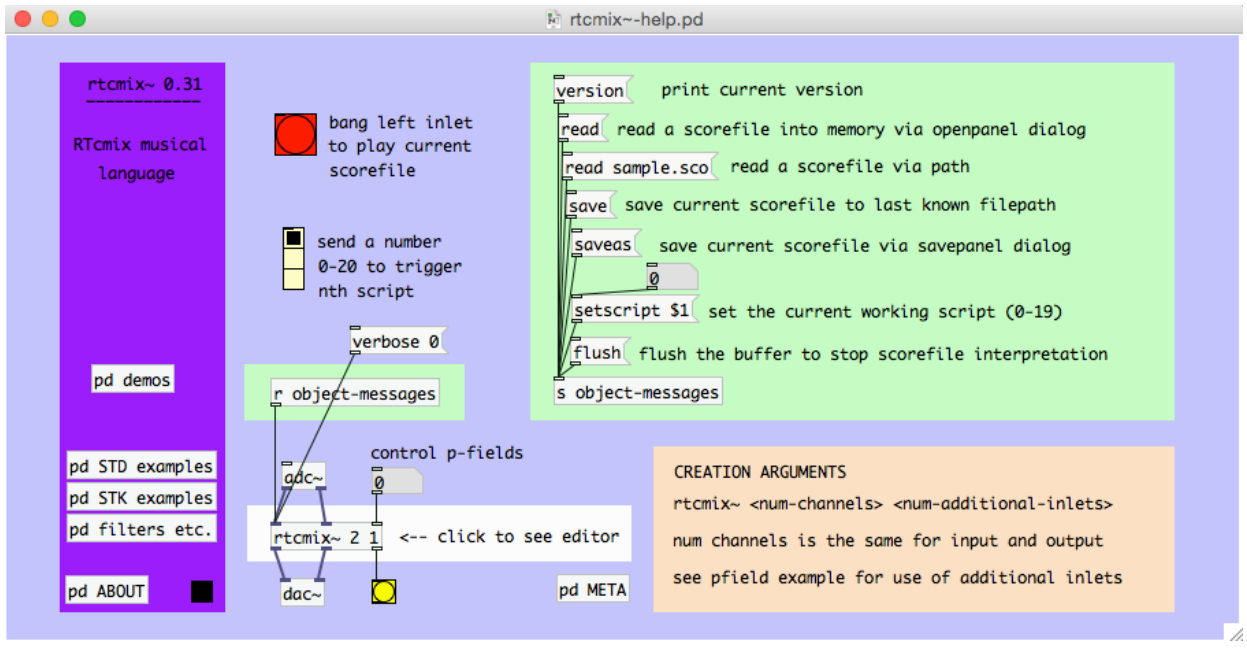

Figure 14: RTcmix's Pure Data object's help window

From here, you'll be confronted with the object's help file, which will introduce you to a number of options for using [rtcmix~] successfully. We're not going to cover all of these options verbatim quite yet, but suffice it to say that now that we are in the visual programming paradigm, you should feel free to click away on any of the messages or objects that are posted and let the tutorial presented in the help file answer any pressing questions that you might have at this point.

For now, we're going to shift our focus to Max/MSP, so go ahead and open Max 7.

Type command-n and you'll see that the presentation for Max is quite a bit different than Pd. Max opens a patcher that is full of a variety of toolbars on each side of its window.

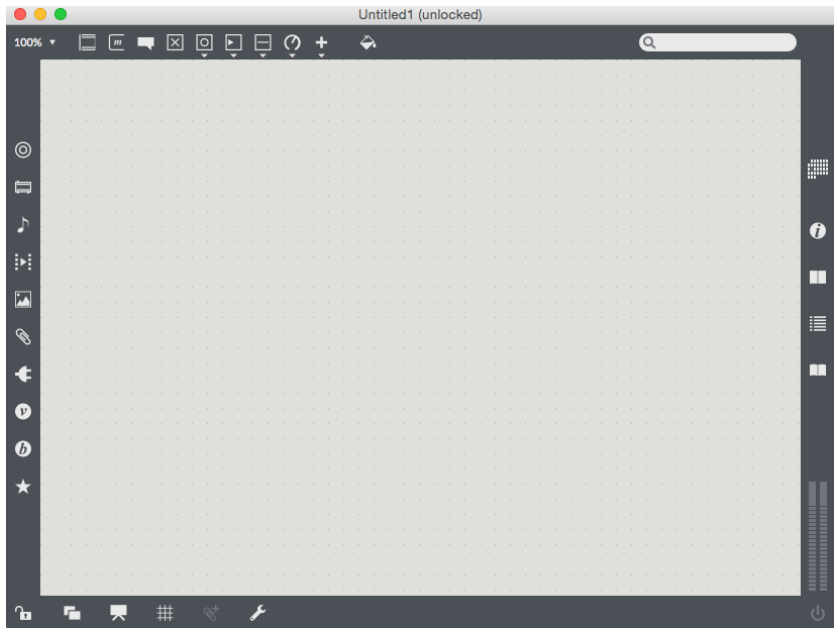

Figure 15: Max 7 patching window

We won't of course outline these tools in detail, but know that hovering over them will indicate, very briefly, what each of them are and for the curious, the little icon that looks like an open book on the right side of the window is a great place to start for mini-lessons about Max itself. I might add, however, that typing command-m will access the Max window, which prints errors the same way that the Pd window does, which can be extremely helpful for debugging your code. You'll see that this window is also accessible by clicking on the icon above the open book that we just mentioned on the righthand side of the patcher window.

The methodology for Max is exactly the same as it is for Pd. We're going to create a variety of patchable objects and messages to realize our work in conjunction with RTcmix. Let's start by rebuilding our basic Pd patch that will play RTcmix score files.

To create an object in Max, type n. You'll create messages by typing m. By typing command-e, you'll toggle between edit mode and run mode, which is verified by the little lock icon on the bottom of your patcher window.

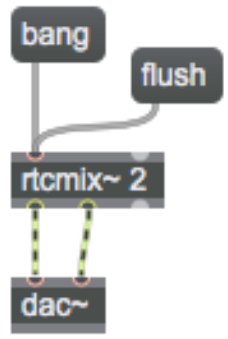

Figure 16: Figure 13 in Max

As we did earlier, click — well, double-click this time — on the [rtcmix~] object while in run mode and a scripting window will open. Type or copy/paste some MINC code into that window and close it to load the code into the [rtcmix~] object. To get sound, click on the power button in the lower right corner of the patcher window and when you're ready, click on the "bang" message to hear your work!

So far so good! Now while in edit mode, right-click on the [rtcmix~] object and open up its corresponding help file.

> No sound? Head to the Menu and navigate to Options → Audio Status and make sure you are processing

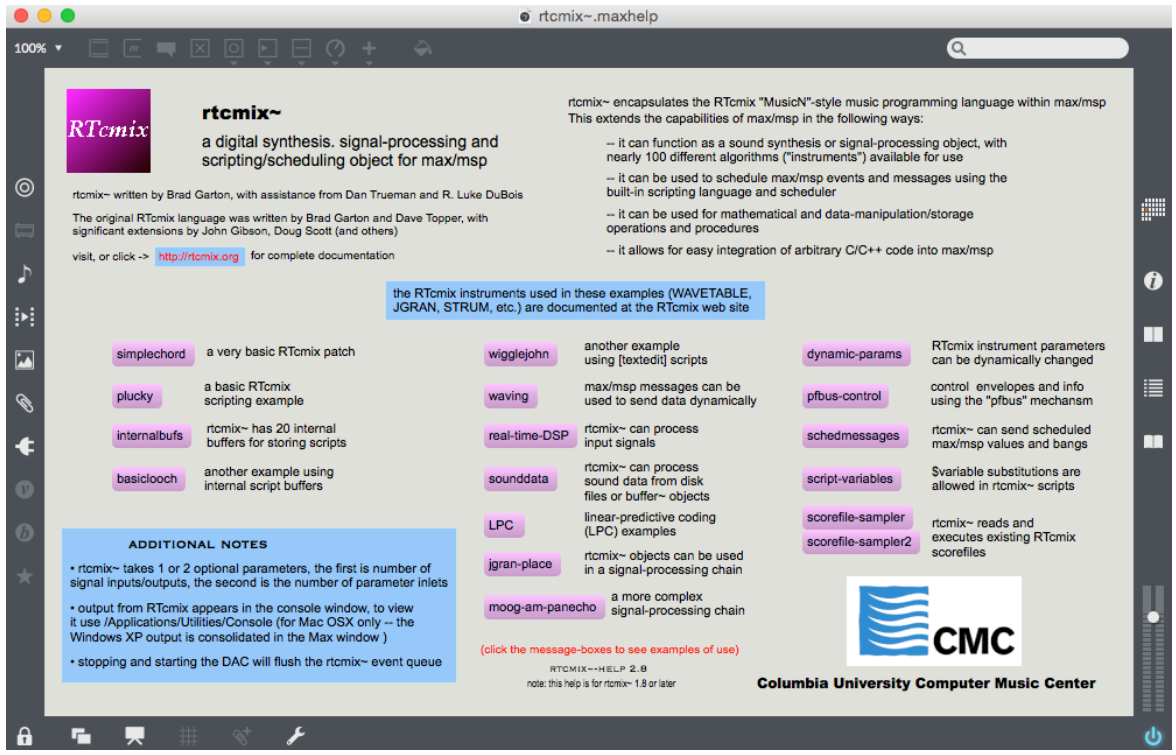

Figure 17: RTcmix's Max object's help file

I highly encourage you to visit each of the modules contained within the help patch and for now, we are going to focus on the important concept of internal buffers.

Until now, we've been utilizing the [rtcmix~] in a singular fashion. That is, we've only loaded one script at a time into its buffer, which has the ability to recall up to twenty buffers at a time. Think of it this way: You'll be able to save up to twenty score files into one [rtcmix~] object at any given time! After we understand how to implement this functionality, we'll see why it is useful for our work.

Start by creating a new message in your existing Pd or Max patch, called "setscript 0" and then connect it to the leftmost inlet of the [rtcmix~] object. This way, we've stored our current score file — the innocuous sine wave or something that you copied and pasted — to buffer 0. Then, create another message, this time called "goscript 0" that will play buffer 0 in just the same way that "bang" did earlier.

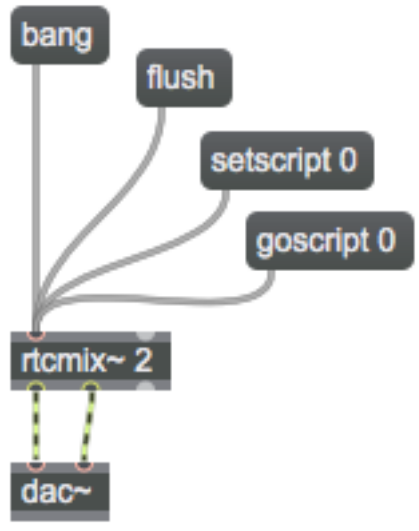

Figure 18: Using "setscript" and "goscript"

In run mode, first click on "setscript 0" and then click on "goscript 0" and you should hear your score file again. Double click on the [rtcmix~] object to reopen the scripting widow and create a new script within it and when you are finished, close the window. Click "bang" and you'll hear your new script, which we are going to store into a new buffer. In edit mode, change the message for "setscript" so that it reads "setscript 1" and then click on it while back in run mode. Then, create a new message called "goscript 1" and connect it to the leftmost inlet of [rtcmix~].

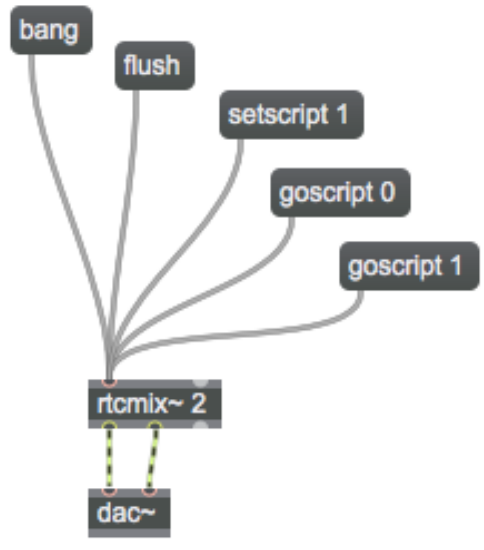

Figure 19: Multiple "goscript" messages

We're now able to click on "goscript 0" to hear our original score file and "goscript 1" to hear the newest script.

> It's customary to utilize comments in your patch window to help you remember what it is that you did while

To better answer the question of why storing multiple buffers is useful or why, even, we're bothering to play back RTcmix scripts in a new paradigm when we've been doing just fine up until now, we need to understand why Pd and Max are useful and why they are so popular in the world of computer music.

We've yet to touch on the notion that while useful as a sound generating platform, we can — and most definitely should — present our work in RTcmix in a live performance setting. While the world of electronic music and more specifically electronic music using computers is relatively young, it is today very common to see and hear live performances where laptops and speakers are just as valid an expressive medium as the piano or violin. While my undergraduate days included an incredible number of hours on my trumpet, when people invariably ask me, as a musician, "So what do you play?" I always respond with, "Laptop computer."

Pd and Max are beautiful programs to be sure, but one of their more outstanding features is their ability to assist you with transforming your laptop into the custom instrument that you might be looking to establish. For example, I've been typing and typing and typing away to create this book, but what if I used all of these keys on my keypad to trigger notes or events, just the same way that piano keys trigger various pitches. Could I create a basic instrument that will allow me to control the [rtcmix~] object in a meaningful way? Well, Pd and Max have a rich array of powerful objects to assist you with realizing your live work — or any work, for that matter — and let's get to answering the above question using a few of them.

Somewhere above all of the code in your patcher window, create a new object called [key] and leave room to create a few more items below it. Now, underneath that [key] object, create a number box by clicking command-3 in Pd or i in Max.

> Quick side note. Max differentiates between floating-point numbers and integers, so always be cognizant of this

Connect the outlet of [key] to the inlet of the number box, get back into run mode, and type the numbers one through zero at the top of your keyboard. You'll notice that each key has a corresponding number coming out of the [key] object, which we're going to use to trigger the various messages we've already written into our patch.

Create a new object, [select], as well as a number of arguments to correspond with the numbers that are generated by the [key] object. When [select] receives a number in its inlet, it produces a bang — which we can think of as a trigger or onset that will do something useful for us — to its corresponding outlet. For example, I would like to trigger my first score using the "1" on my keyboard and the second score file using the "2." Therefore, I'm going to need to use [select 49 50] as my object with corresponding arguments. I'll take it a step further, however, and include the space bar for playing the currently stored buffer and the delete key for stopping score file rendering using the "flush" message.

> Those of you working in Pd will note that the delete key corresponds with 8 and not 127 as it does in Max.

Figure 20: Using the [key] and [select] objects

You can imagine the versatility that we can create with this patch simply by storing enough score file buffers into our object to correspond with the top row of numbers on our keyboard. If you'd like to play with more than twenty buffers, you'll need to create a new [rtcmix~] object that can be connected to the same [dac~] object and filled with its own scripts.

> In edit mode, you can highlight entire portions of your patch by clicking, holding, and dragging to surround

Both Pd and Max have their own set of powerful signal processing objects and you'll greatly enhance your experience with these programs when you utilize their DSP tools in conjunction with RTcmix. Earlier we discussed the notion of ring modulation, which is very easy to implement in Pd or Max and can add another layer of complexity and flexibility to our patch.

We'll transition back to Pd, but again, all of this can be done in Max using the same objects and number boxes, albeit with the caveat of specifying floating-point numbers or integers.

Disconnect the patch lines from the outlets of [rtcmix~] and [dac~] and create a bit of space between each of them. We're going to add the ring modulation

effect after [rtcmix~] using the same methodology that we are familiar with using bus_config( ), albeit with some enhanced control thanks to our visual programs. Recall that ring modulation multiplies a complex incoming signal with a simple modulating signal to produce sum and difference tones, or sidebands.

Both programs include a suite of tools for arithmetic and you can work with control data or signal data. Any object with the "~" following it will manipulate signals, which is evidenced by the look of their outlets and patch lines. Create a [*~] object to multiply signal values and connect one of the outlets of [rtcmix~] into its leftmost inlet. Next, you'll create the [osc~] object in Pd or the [cycle~] object in Max to create a sine wave oscillator. Connect your sine wave to the rightmost inlet of [*~] to complete the multiplication. Then, connect a number box to the inlet of your [osc~] or [cycle~] object. Finally, connect the outlet of [*~] to both of the inlets for the [dac~] object.

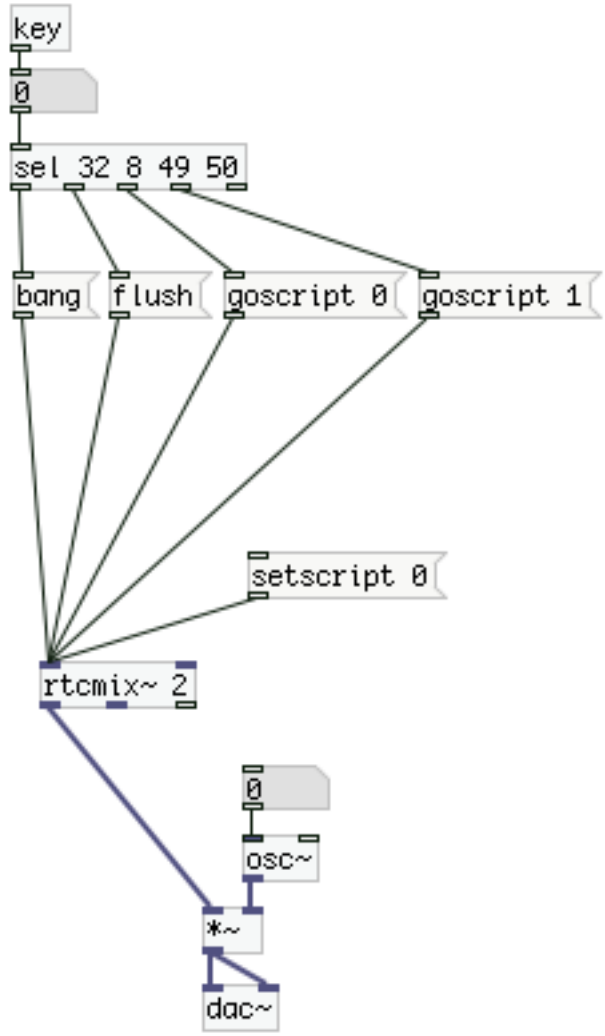

Figure 21: Adding ring modulation

Double click on a number box, type in the desired value (in this instance, the frequency of our oscillator), and hit enter. This will store that value to our oscillator, but we can also click on the number box, hold, and drag up and down to sweep through our varying sidebands. Pretty fun!

Suffice it to say that with some research and practice, you'll quickly gain a great deal of facility on Pd and/or Max, which will be especially useful for your work with RTcmix. These two programs greatly enhance the live performance experience, as they are particularly adept at transforming your laptop into a completely customizable instrument. Moreover, they are incredibly facile at controlling MIDI data, so mapping your new USB keyboard or control surface is a cinch.

> It should be noted, however, that RTcmix contains this functionality via makeconnection("midi"), which we'll

Our next Interlude will continue our work with Pd and Max and the ways in which we can use their objects and messages to control p-field data in our RTcmix scripts in real time as well as ways that we can incorporate acoustic instruments into the fold for potential electroacoustic works.

Sonata VII: Pd or Max patch

Using the principles outlined in this Sonata, create a Pd or Max patch that includes at least three stored buffers and is mapped to your keyboard in a way that spells out your initials or first name. Thus, it will be possible for someone to type your initials or name into the patch and hear your corresponding RTcmix sounds in a meaningful, well-sculpted musical etude.

|| Interlude VII: Interactivity ||

We've yet to explore the critical world of interaction with live, acoustic instruments using RTcmix. With our newfound facility in Pd and Max, we'll be able to take live input from a microphone and process it via the [rtcmix~] object, which can be controlled or further processed, depending on our desired implementation. We're going to learn how to capture audio and update p-field parameters from Pd and Max, and finally utilize all of our tools to create a basic, performance-ready patch.

Earlier, we were introduced to the [dac~] object, which translates digital data to analog signal. The [adc~] object does the opposite: It takes real-world acoustic energy captured by your microphone and translates it to streams of digital data for processing in our patches. Start with a new Pd or Max patch and create an [rtcmix~] object that is ready for stereo output. Then, create an [adc~] object and connect its leftmost outlet to the leftmost inlet of [rtcmix~]. Finally, create a [dac~] object and connect the outlets of [rtcmix~] to [dac~] the same way we did earlier.

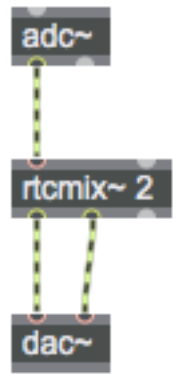

Figure 22: Audio input to the RTcmix object

We'll also need "bang" and "flush" messages to play our score file once we have sound ready to process.

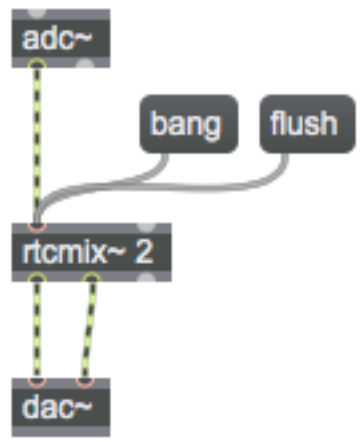

Figure 23: Adding "bang" and "flush" messages

You are free to code your score files in your preferred text editor, or you can write them directly in the scripting window that pops up once you click on the [rtcmix~] object. I will say, however, coding in this window runs the risk of losing your work should Pd or Max crash.

Up until now, we've limited our incoming audio to sound files accessed via rtinput( ) pointing to files on our computers. However, now that we are in the

Pd/Max paradigm, we can utilize rtinput("AUDIO") to capture audio that is incoming via the [adc~] object. Remember that we don't need to call upon rtsetparms( ) or load( ) while using the [rtcmix~] object, so your first line in your score file can be a call to rtinput( ).

```
rtinput("AUDIO")
```

Depending on how you've configured your system in Pd or Max to handle incoming sound, you'll now be able to process live audio. If you haven't already done so, double check your Audio Settings (Pd) or Audio Status (Max) to make sure that incoming sound is coming from your internal microphone or external, standalone audio interface with a microphone connected to it.

We'll begin with a simple processing example, so let's add a bit of delay to our voices as we speak into the microphone.

```
rtinput("AUDIO")
start = 0
instart = 0
duration = 30
amplitude = 1.0
delay_time = 1.0
r_ch_amplitude = 1.0 // right channel amplitude relative to left channel
DEL1(start, instart, duration, amplitude, delay_time, r_ch_amplitude)
```

Score file 54: Simple delay of incoming signal

Before closing the scripting window and clicking on the "bang" message to start processing sound, make sure that you are processing audio by checking the DSP button in Pd or the power button in the lower right corner of your patching window in Max. Also, be sure to wear headphones or the proximity of your internal microphone to your internal speakers will onset some wicked feedback that we don't want.

If all goes well, you should be able to hear a delayed copy of yourself as you speak or sing into your microphone!

RTcmix will of course stop processing audio after 30 seconds, but what if we'd like to use this budding Pd or Max instrument's functionality indefinitely? While we could amend our duration within the script itself, we can call upon MAXBANG( ), which outputs a bang to the rightmost outlet of [rtcmix~] at a specified time. For example, we'll amend score file 54 by indicating a duration of five seconds. We'll also use MAXBANG( ) to output a bang after five seconds, which we can then use to loop our score file.

So, using score file 54, add a new line at the end of the script and add the following command:

```
MAXBANG(5)
```

Then, you'll need to add an object coming out of the right outlet of [rtcmix~] called a bang button, which you can create by typing shift-command-b in Pd or b in Max. The outlet of this bang button is connected to the inlet of the "bang" message above, which will create a loop. After clicking on the original "bang" message, our score file will loop indefinitely, which we're going to want some control over.

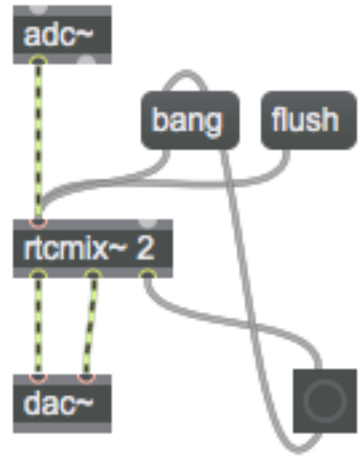

Figure 24: Looping our scripts

We don't really have a way to stop this loop from continuing, so for now, head back to edit mode and disconnect the patch line from the bang button to and add the following to your patch, depending on if you are using Pd or Max, respectively.

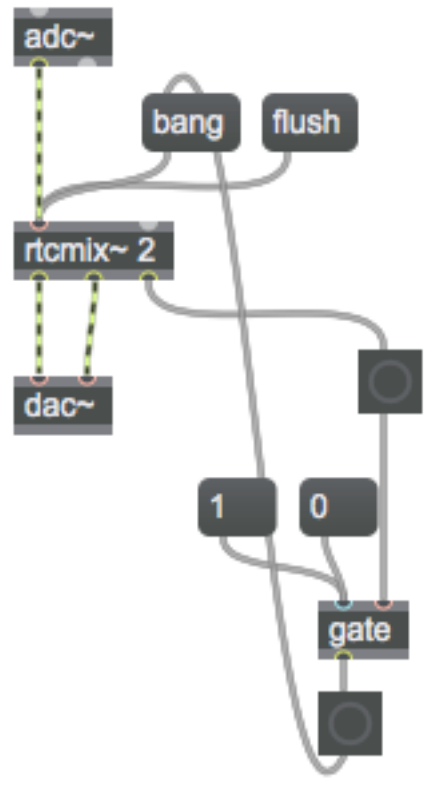

Figure 25: Adding [gate] in Max

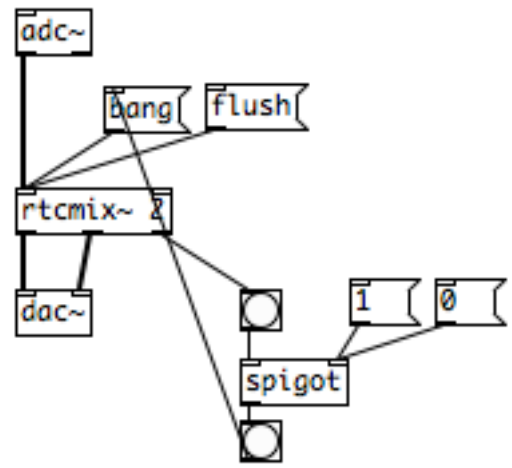

Figure 26: Adding [spigot] in Pd

The [gate] object in Max and the [spigot] object in Pd are aptly named. Note how each respective object takes a bang button and the messages "0" and "1" into its inlets, though they are reversed depending on your program of choice. Functioning like a gate or spigot, these objects will allow data to pass through it as long as "1" is selected, and will stop data from passing with a "0" message. In this way, we can control whether or not our [rtcmix~] object will run indefinitely.

We're familiar with updating p-field parameters using table data, however the [rtcmix~] object can be further enhanced to include a multitude of inlets that will accept control data from Pd or Max to pass to the internal p-fields. We're going to need to create a new [rtcmix~] object with two arguments: "2" for the number of outlets that we'd like to use, and "1" for the number of inlets that we need. Moreover, let's connect a number box to the rightmost inlet of [rtcmix~] and be sure to create it using f in Max for floating numbers.

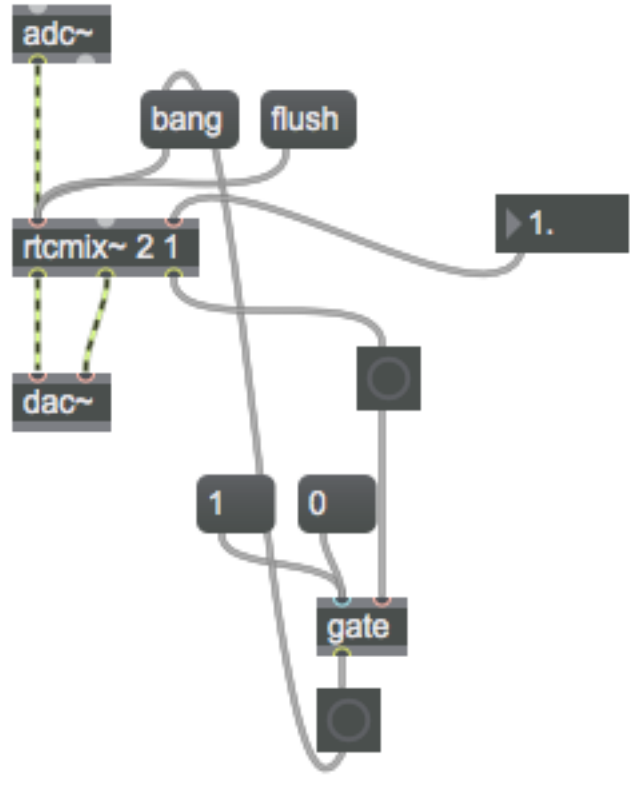

Figure 27: Adding an inlet for p-field updating

This new inlet can receive streaming numbers from Pd or Max by way of the makeconnection("inlet") command. For example, we're going to control our delay time using this inlet and including the following in our score file.

```
delay_time = makeconnection("inlet", 1, 1.0)
```

We are now able to transition back to run mode and interact with our delay times. For example, any number box can be altered by first double clicking on it, typing in your desired number, and clicking enter. Or, you may click on the box, hold, and scroll up and down to dynamically update values. If, on the other hand, you have a set series of numbers that you'd like to utilize as delay times, please

> Amend your MINC code to have a duration of 30, as well as MAXBANG(30) so that we can truly hear

feel free to create a set of messages that connect to the requisite inlet of the [rtcmix~] object.

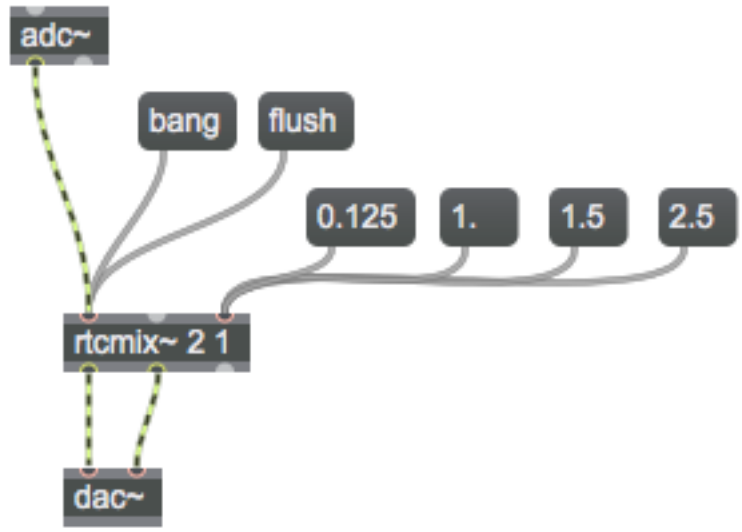

Figure 28: Several messages for p-field updates

We could use Pd or Max to generate random numbers to use as dynamic updates for any of our p-fields that use makeconnection("inlet") using the [random] object. Carefully reconstruct the following snippet of code and prepare to connect it to the rightmost inlet of your [rtcmix~] object.

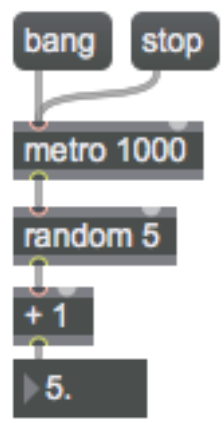

Figure 29: Generating random numbers

Starting with two messages, "bang" and "stop," we are able to control a metronome using the [metro] object. Time in Pd and Max is calculated in milliseconds, so initializing a [metro] object with 1000 will ensure that a bang is sent out of its output every second. The [random] object accepts bangs into its inlet and once it receives a bang, outputs a random number between zero and one less the number specified in its creation argument. In this case, I'm outputting numbers between one and four because I've added a [+] object to add one to each incoming number from [random]. Thus, my final range of values will be between one and five.

By turning this metronome on, we can speak or play into our microphones, continually loop our [rtcmix~] object to render indefinitely, and interact with random delay times.

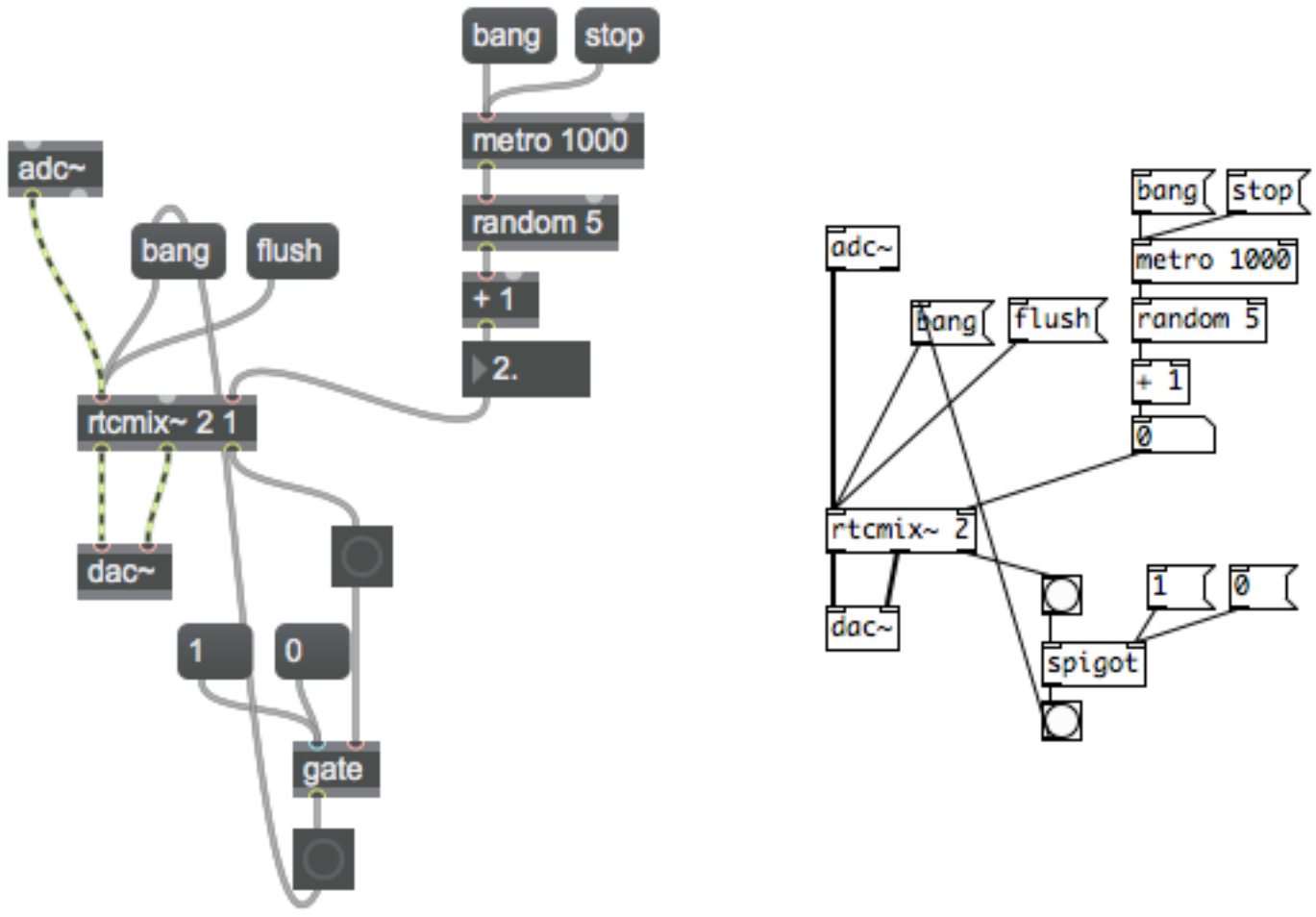

Figure 30: Finished code in Max and Pd, respectively

If this is your first time using Pd or Max, I'd like to propose a small challenge. Now that we have some practice with the methodology of connections in these two programs, see if you can devise a way that will start your [rtcmix~] script, open your gate or spigot, and start your metronome all at once using the space bar. Then, map your delete key to turn all of those elements off at once.

We're going to shift focus and concentrate instead on a powerful way to interact with the audio that is incoming via the [adc~] object. One of the most powerful objects in both Pd and Max is [sigmund~], a real-time audio analysis tool that can return many important strings of data, including pitch.

If you'd like, go ahead and open a new patch window and save it to your folder of working Pd or Max patches. Then, code together this small snippet.

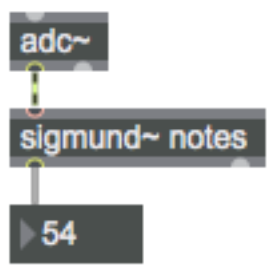

Figure 31: Incorporating [sigmund~]

Once you have this patched together, make sure that you are set to compute audio and sing a few pitches into your microphone. If all is working correctly, you should see some changes and know that [sigmund~] is tracking your pitches in real time and outputting them as values to your number box!

Now, we specified notes as an argument for [sigmund~], which outputs MIDI note values and only shifts values once a change in pitch is interpreted. Change

the creation argument from notes to pitch and observe the difference in the stream of values.

> For those who might use or are exploring the pitches argument, take note: When no sound is produced and ana-

Let's revert back to notes and ensure that our values will be rounded up to the nearest integer value. We'll also translate those MIDI note values to Hz using the [mtof] object.

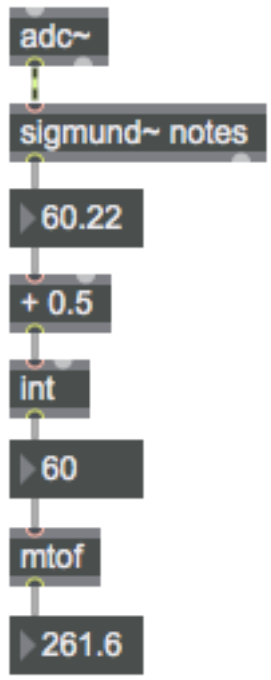

Figure 32: Rounding our values and translating to frequency

We'll also create an [rtcmix~] object and a [dac~] object and connect them accordingly. As a proof of concept, we're not going to connect the [adc~] object to [rtcmix~] just yet.

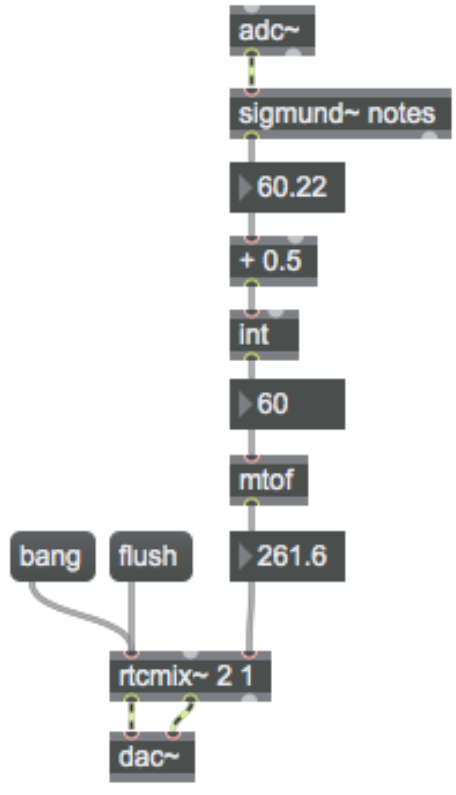

Figure 33: Adding RTcmix

Turning our attention back to the MINC code within our [rtcmix~] object, we're going to introduce a new instrument, MBLOWBOTL( ), which we'll control using our pitch tracker.

```
start = 0
duration = 30
amplitude = 30000
pitch = makeconnection("inlet", 1, 261)
noise_gain = 0.25
pressure = 0.75
MBLOWBOTL(start, duration, amplitude, pitch, noise_gain, pressure)
```

Score file 55: Controlling MBLOWBOTL( ) within [rtcmix~]

MBLOWBOTL( ) is a member of the suite of STK instruments in RTcmix and replicates the sound produced by blowing over the top of a bottle. Note that sound is continuous and updates only when we sing or speak into our microphone because we are using the notes argument in [sigmund~].

Connect the outlet of [adc~] to the leftmost inlet of [rtcmix~] and clear your scripting window so that we can explore live processing of our sounds via VOCODESYNTH( ), which is one of RTcmix's vocoders. The vocoder (voice encoder) was originally developed as a means of transmitting the human voice over telephone lines and is a popular synthesis technique for realizing interesting sound effects and is a common component in analog electronic music. A vocoder generates sounds based on the spectral characteristics of an input signal that is first analyzed and then synthesized using any of a number of filters.

```
rtinput("AUDIO")

start = 0
instart = 0
duration = 30
amplitude = 10

filters = 40
low_center_frequency = 220
interval_spacing = cpspch(0.02) / cpspch(0.0)
transposition = 0.00
bandwidth = 0.009
window_length = 0.001
smoothness = 0.98
threshold = 0.0001
attack = 0.001
release = 1.0
high_mix = 0.25
high_center_frequency = 2000

input_channel = 0
```

```
pan_frequency = makeconnection("inlet", 1, 5.0)
pan = makeLFO("sine", pan_frequency, 0,1)
carrier_wave = maketable("wave", 1000, "tri")

scalecurve = maketable("curve", "nonorm", 1000, 0.0,0.25,1.0, 1.0,1.0)

VOCODESYNTH(start, instart, duration, amplitude, filters, low_center_frequency,
          interval_spacing, transposition, bandwidth, window_length,
          smoothness, threshold, attack, release, high_mix, high_center_frequency,
          input_channel, pan, carrier_wave, scalecurve)
```

Score file 56: Robot voice

Before running this script, let's cover the constituent parts of VOCODESYNTH( ) and amend our Pd or Max patch a bit to best utilize our interaction with makeLFO( ).

We're using 40 filters in our sound, though you are of course free to play with that value and make note of the changes. After setting the center frequency of the lowest in the set of filters, we specify how the remaining center frequencies are determined, intervalically. In this instance, we are dividing our interval (cpspch(interval)) by cpspch(0.0), which will return stacks of the interval using equal temperament. You'll then see that we are not transposing our incoming sound and are using very minute values for our filter constructs.

The real interaction occurs in our p-field for panning. Using our makeLFO( ) command, we can dynamically update the frequency of our low frequency oscillator using makeconnection("inlet"). Thus, as you sing or whistle or speak pitches that are higher into your Pd or Max patch, the frequency for our LFO will change accordingly.

In order to get pitches that are less than 20 Hz — which we want in order to truly be low frequencies — we need to divide our outgoing pitches from [mtof] by ten, or more depending on the range of your voice.

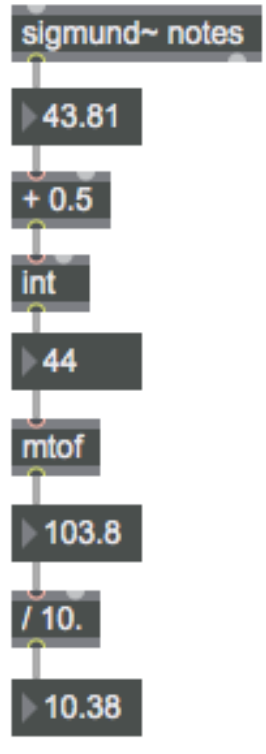

Figure 34: Division for LFO

Make note that if the frequency for makeLFO( ) reaches to or exceeds about 20 Hz, the LFO itself becomes audible and will modulate our vocoder sounds for yet another level of complexity.

We've merely scratched the surface of the multifarious possibilities afforded to anyone who is interested in composing interactive music using RTcmix in conjunction with Pure Data and/or Max/MSP. As noted earlier, Max 7 includes a wide array of learning tools built right into the program, including videos, tutorials, and of course help patches. For those who'll continue with Pd, the FLOSS (Free Libre Open Source Software) Manual is a tremendous resource, which you can find at http://en.flossmanuals.net/pure-data/. Moreover, while the site is

about to become obsolete in April of 2016, I've found the Max Objects Database (which includes objects for Pd as well) to be very helpful in finding new objects that are written by the community of users and might not come packaged in Pd or Max. Check out http://www.maxobjects.com for as long as the site is up and running.

Interlude VII: Interactive Pd or Max Patch

Compose an improvisatory work for your instrument or voice that uses pitch tracking to manipulate a p-field in your RTcmix script. You are free to make your piece as simple or complex as you like, though exploring graphic notation for your instrumental part is of course welcomed, as is using an array of RTcmix instruments via bus_config( ).

|| Postlude: Customization ||

RTcmix is a totally customizable experience and can be tailored to your individual tastes and needs. If we look all the way back to our work with installation, we noticed that there are three different ways to use the program. Moreover, some of you might have been using RTcmix in conjunction with Python for our entire journey together, rather than the standard MINC parser. Our work just took us through using RTcmix in conjunction with Pure Data and Max/MSP. In short, RTcmix shouldn't be thought of as software in the same way we view Propellerhead's Reason audio workstation, for example. Unlike proprietary software, we don't need to wait for the next X.x release to add to our enjoyment or finally include that bit of functionality we've been craving all along. Rather, RTcmix affords us the ability to use it on our own terms, using only, or adding to, the features that we want to utilize. At its heart, RTcmix is a sophisticated suite of instrument libraries and commands that will assist us in realizing our music, whatever the platform may be.

We're going to wrap up our work with RTcmix by briefly introducing a few of the customizable features of the program itself. We'll begin by looking at some of its graphical possibilities via some ancillary downloads, including printing tables and using the mouse trackpad as a data source. Then, we'll explore MIDI and how we can use USB MIDI controllers in conjunction with RTcmix.

Earlier, we introduced the concept of envelopes — the ADSR envelope in particular — and included a few graphs to help us get a sense of each envelope's shape. This was especially useful when considering the difference between creating tables using maketable("line") and maketable("curve"). In order to crate you own graphical tables using RTcmix's plottable( ) command, we're going to need a few ancillary downloads.

First, we'll download AquaTerm, which is a graphics application for OS X and will be our vehicle for viewing graphs and tables in RTcmix. Visit http://sourceforge.net/projects/aquaterm/files/?source=navbar and click on the link for the .dmg image. Double click on the .dmg and run the package installer, which will place AquaTerm in your Applications folder.

Next, we're going to download an OS X package manager called Homebrew, which will be the easiest way for us to install a package called gnuplot, which will complement our AquaTerm download.

Type the following into a new Terminal window. The installation process might take a bit, so hang tight.

> You can also visit http://brew.sh and follow the directions from their website.

```
/usr/bin/ruby -e "$(curl -fsSL https://raw.githubusercontent.com/Homebrew/install/master/install)"
```

Lastly, we're going to "brew" gnuplot using the Homebrew package manager. You can do this by typing the following line into Terminal.

```
brew install gnuplot --with-aquaterm
```

With this, we're now all set for including graphics and plotting tables with RTcmix! Check your work by opening score file 15 and adding the following to your script.

```
plottable(envelope)
```

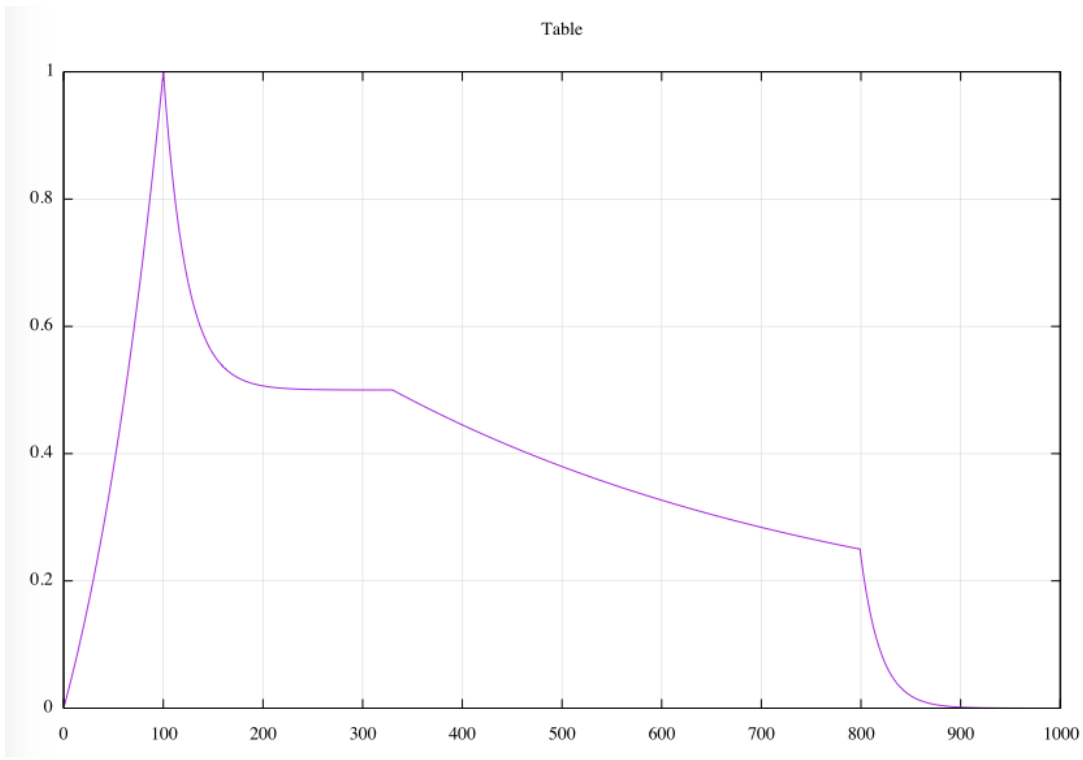

Figure 35: Curvy ADSR from score file 15, viewable using plottable( )

Not only can we plot tables, but the new graphics window will allow us to use our trackpad as a source for real-time data via makeconnection("mouse").

```
rtsetparams(44100, 2)
load("STEREO")
rtinput("/Users/jerod_s/Desktop/organ-test.aif")

start = 0
instart = 0
duration = DUR()
amplitude = makeconnection("mouse", "y", 0,1, default = 0.5, lag = 50)
```

```
pan = makeconnection("mouse", "x", 1,0, default = 0.5, lag = 50)
STEREO(start, instart, duration, amplitude, pan)
```

Score file 57: Using data from trackpad

Amplitude data is tracking on the y axis and pan data on the x axis of our mouse window, which you can see is set in the second p-field of makeconnection("mouse").

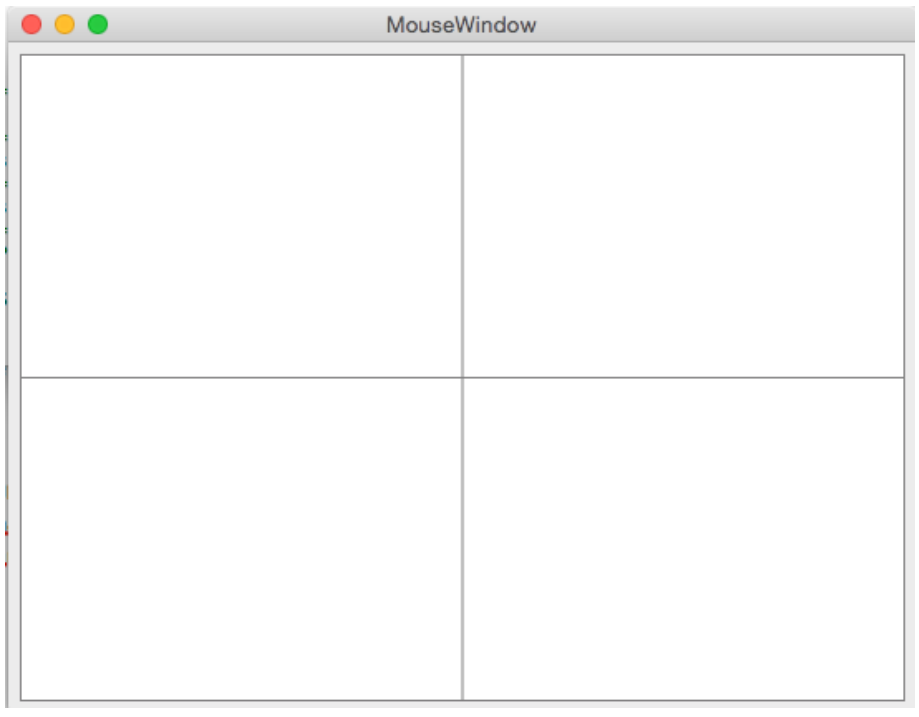

Figure 36: Mouse window

From there, we set a range of values — written in pairs — as well as a default value to use when starting our script, a lag time in milliseconds to smooth out changes in values. If we didn't specify a lag time, we'd hear clicks and pops as we drag our cursor around our window.

For many, the use of USB MIDI controllers is a way to enhance the experience of playing live, due to the array of buttons, knobs, and sliders that can be utilized to alter data in their program of choice. We can use MIDI data with RTcmix in two ways and we'll first explore MIDI control in Pd and Max, then look at MIDI control natively in RTcmix itself.

My current controller of choice is the Livid block, which is made by Livid Instruments, though, sadly, the unit is now out of production. I enjoy using this particular controller because of its sheer number of buttons (64 in all), which I can use to trigger events in RTcmix, along with its corresponding row of eight knobs.

Any knob, button, or slider has a corresponding value associated with it, much like we observed with the [key] object in Pd and Max. Thus, it will be possible to select any of those values for use in our patch and then passed to the [rtcmix~] object via makeconnection("inlet").

Take a moment to ensure that your USB-connected MIDI controller is hooked up to your computer and then open Pd or Max. In Pd, we're going to need to visit the Menu bar and select Media → MIDI Settings. From there, you'll be able to select your input and output MIDI devices. In this case, I'm looking for my block controller and once you find yours, click on "Apply" and then "OK."

If you are using Max, head to the Menu bar for Options → MIDI Setup and select your controller.

We aren't going to be concerned with passing MIDI messages out of Pd or Max, so it's not imperative to select your controller as an "Output" device, but it's a good habit to do straight away as Pd and Max will save your settings the next time you open the program, in case you do eventually want to pass MIDI messages out to an outboard synthesizer, for example.

Depending on your program of choice, take time to carefully code out the following in a new patching window and pay particular attention to the subtle differences in the Max patch, such as the need to specify floating point number boxes.

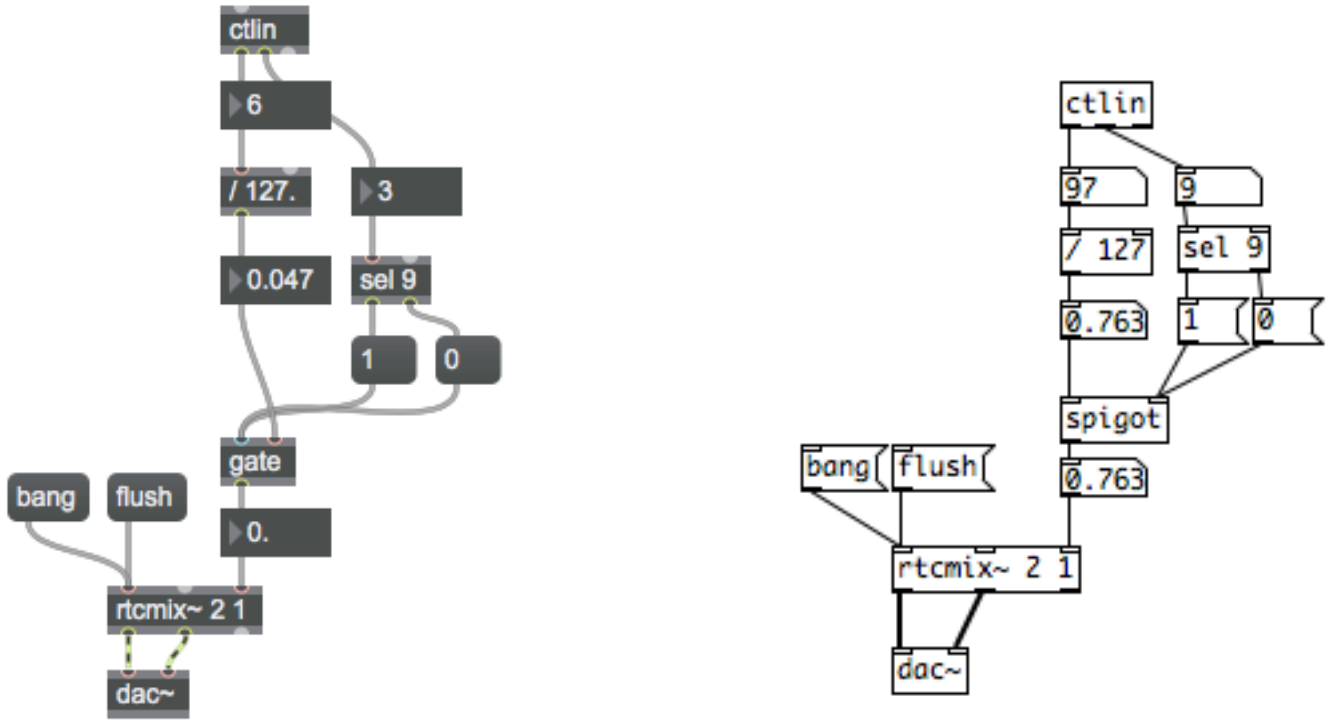

Figure 37: MIDI control in Max and Pd

The [ctlin] object outputs three numbers. The leftmost outlet is the actual control value, which is a stream of numbers from 0-127 that corresponds with the range of numbers in the Musical Instrument Digital Interface (MIDI) protocol. The middle outlet references the control number, which in this case is the far left slider toward the bottom of my Livid block. Each time I move that slider, [ctlin] will output a "9" as that is the referential number associated with that particular slider. The far right outlet corresponds with the MIDI channel assignment, which we aren't concerned quite yet.

The range 0-127, while useful, can be amended. In this case, I'm dividing that range of numbers by 127, in order to put them in the range of 0-1. That

way, I can use them for, say, amplitude values in my script. Moreover, the [sel] object (short for "select") will ensure that my [gate] or [spigot] is open only when I move the slider in question.

Here is a simple score file for testing this patch now that we have control over at least one aspect of our script.

```
rtinput("/path/to/file.aif")

start = 0
instart = 0
duration = DUR()
amplitude = makeconnection("inlet", 1, 0.0)
pan = 0.5
STEREO(start, instart, duration, amplitude, pan)
```

Score file 58: Amplitude control using MIDI data from Pd or Max

The only real advantage in constructing our MIDI mapping in Pd or Max is of course the chance to add any Pd or Max-specific elements to our work. RTcmix is equipped with the ability to make care of its own MIDI work in-house, so to speak, using set_option( ) and makeconnection("midi").

```
rtsetparams(44100, 2)
load("STEREO")
rtinput("/path/to/file.aif")

set_option("midi_indevice = block")

default = 0
lag = 25
channel = 1
```

```
slider1 = makeconnection("midi", 0.0,1.0, default, lag, channel, "cntl", value = 9)
STEREO(start = 0, instart = 0, duration = DUR(), amplitude = slider1, pan = 0.5)
```

Score file 59: MIDI input using makeconnection("midi")

In order to correctly identify your controller for the set_option("midi_indevice") command, you'll need to open your Audio MIDI Setup application, which is found in your /Applications/Utilities folder. Once you have that opened, click command-2 to open the MIDI setup window, which will list your controller.

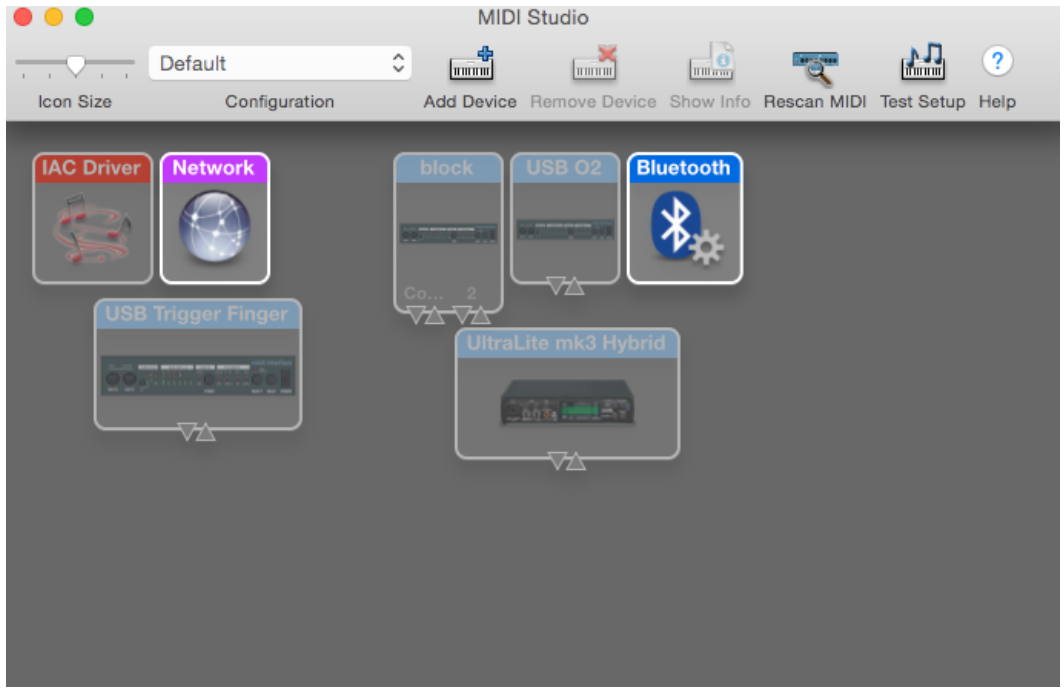

Figure 38: Using Audio MIDI Setup

In my case, my computer recognizes my controller as "block," so I'll change that accordingly in my call to set_option( ). I could use my M-Audio Trigger Finger controller by declaring set_option("midi_indevice = USB Trigger Finger").

We will need to know which channel our controllers are sending MIDI data through, which you can find by either clicking on your controller's icon in Audio MIDI setup or checking our the number from the rightmost outlet of [ctlin] in Pd or Max.

You'll see that we're creating a connection to the slider1 variable, which includes a desired range of values (0.0 - 1.0), a default value, lag time, channel number, the type of data that we're looking for, and the corresponding number associated with our slider.

This block controller has 64 buttons that I also use in performance. These buttons are treated similarly to piano keys on a MIDI keyboard controller, in that they correlate with pitches and are different from control data. In Pd or Max, you can get data from these buttons or keys by using the [notein] object and in RTcmix, you'll need to specify "noteon" instead of "cntl" in p-field six.

It can't be stressed enough: RTcmix is a completely customizable suite of open source tools. If there's one overarching mantra of this book, it's that you are free to use RTcmix in any way that you wish. It plays well with MINC, Python, Pd, Max/MSP, MIDI controllers, and even iOS and Android. Constantly evolving, it is curated by a strong community of users and continues to receive upgrades, enhanced functionality, and exciting new features that will ensure it continues to be a powerful choice for electronic and computer musicians for years to come.

|| 100 score files ||

The following examples were copied, culled, and edited from a year-long project I set forth for myself in 2013, aptly called *Script Calendar*. For the entire calendar year — in an effort to truly understand RTcmix from a variety of angles — I wrote a complete score file for each day. It was a wonderful, personal exploration and one that I am proud to have gone through and my hope is that these 100 score files serve you well as you learn the program for yourself.

If this book is being used in conjunction with a semester-long class, then 100 files will more or less provide you with the ability to rewrite one of them each day during the term, with a few days here and there for a day off. I find that like any language or skill, it is only possible to truly learn and gain facility through small, daily practice and immersive, hard work. These score files are now yours, to copy, hack, alter, and sculpt into your own compositions and I sincerely hope that you have as much fun with them as I did when I worked them out myself.

```
//Day 1
print(“Hello World!”)

//Day 2
x = 12.6
printf("the variable x equals: %f, %s \n", x, "C-style printing, diff from using print”)

//Day 3
rtsetparams(44100, 2) //set sample rate, num channels
load("WAVETABLE") //load instrument
```

```
WAVETABLE(0, 5, 3000, 440, 0.5) //start, duration, amplitude, frequency, pan
//Day 4
rtsetparams(44100, 2)
load("WAVETABLE")
envelope = maketable("line", 1000, 0,0, 0.1,0.8, 0.8,0.5, 1.0,0) // add envelope
WAVETABLE(0, 5, 3000*envelope, 440, 0.5) //sine tone

//Day 5
rtsetparams(44100, 2)
load("WAVETABLE")
envelope = maketable("line", 1000, 0,0, 0.1,0.8, 0.8,0.5, 1.0,0)
wave = maketable("wave", 1000, "saw") // sawtooth wave
WAVETABLE(0, 15, 8000*envelope, 440, 0.5, wave)

//Day 6
rtsetparams(44100, 2)
load("DECIMATE")
rtinput(“/path/to/your/file.wav“)
envelope = maketable("line", 1000, 0,0, 0.1,1.0, 0.5,1.0, 1.0,0)
pan = makeLFO("sine", 0.3, 0,1)
DECIMATE(0, 0, DUR(), 0.5, 0.8*envelope, bitdepth = 2, 0, pan)

//Day 7
rtsetparams(44100, 2)
load("WAVESHAPE")
frequency = makeLFO("sine", 0.1, 400,10000)
envelope = makerandom("prob", 10, -1.0,1.0,0,0)
pan = makerandom("prob", 10, -1.0,1.0,0,0) //random panning
wave = maketable("wave", 1000, "sine")
transferfunction = maketable("cheby", 1000, 0.9, 0.3,-0.2,0.6,-0.7)
indexguide = maketable("line", 1000, 0,0, 3.5,1, 7,0)
WAVESHAPE(0, 20, frequency, 0, 1, 8000*envelope, pan, wave, transferfunction, indexguide)

//Day 8
rtsetparams(44100, 2)
load("AM")
rtinput("/Users/jerod_s/Desktop/SC9.aiff")
env1 = maketable("literal", 1000, 0,0, 2,1, 3,0.5, 7,0.5, 1,0)
env2 = makerandom("prob", 10, -1.0,1.0,0,0)
pan1 = makeLFO("saw",  8.5, 0.3, 0.7)
pan2 = makeLFO("buzz", 0.2, 0,   1.0)
wave = maketable("line", 1000, 0,-1.0, 0.5,1.0, 0.51,-1.0, 1.0,0) //ugly sawtooth wave
```

```
AM(0, 0, DUR(), 8.0*env1, 25025, 0, pan1, wave) // aliased carriers
AM(2, 0, DUR(), 8.0*env2, 25050, 0, pan2, wave)



//Day 9
rtsetparams(44100, 2)
load("MBLOWBOTL")
makegen(1, 5, 1000, 0.1, 50, 1, 50, .8, 600, .8, 300, 0.1) //old-style envelope
increment = 1
for(start = 0; start < 50; start += increment){
     amplitude = irand(40000, 80000)
     duration = irand(0.1, 4.0)
     pan = pickwrand(0.0,40, 0.5,20, 1.0,40)//target-value, probability(%)
     frequency = irand(0.1, 18) //subsonic
     MBLOWBOTL(start, duration, amplitude, frequency, noise = 0.3, pressure = 0.5, pan)
     increment = irand(0.1, 0.8)
     }

//Day 10
rtsetparams(44100, 2)
load("WAVETABLE")
load("SHAPE")
bus_config("WAVETABLE", "aux 1 out")
waveform = maketable("wave", 1000, “saw124")
WAVETABLE(0, 30, 10000, 0.34, 0, waveform)

bus_config("SHAPE", "aux 1 in", "out 0-1")
amp = maketable("line", 1000, 0,0, 9,1, 10,0) * 0.25
pan = makeLFO("buzz", 0.4, 0,1)
transferfunction = maketable("random", 10, "triangle", -1.0,1.0)
indexguide = maketable("window", 1000, "hanning")
SHAPE(0, 0, 30, amp, min=0, max=3.0, 0, 0, pan, transferfunction, indexguide)



//Day 11
rtsetparams(44100, 2)
load("AMINST")
control_rate(2000) //reset update rate

duration = 1
amplitude = 8000
```

```
envelope = maketable("window", 1000, "hanning")  //smooth
modulatoramplitude = maketable("line", 1000, 0,0, 1,1, 2,0)
carrierwave = maketable("wave", 1000, "buzz9")
modulatorwave = maketable("wave", 1000, "sine")
carrierfrequency = cpspch(8.00)
modulatorfrequency = cpspch(8.02)
increment = 0.5

for(start = 0; start < 20; start += increment){
     AMINST(start, duration, amplitude*envelope, carrierfrequency, modulatorfrequency,
            pan = 0.5, modulatoramplitude, carrierwave, modulatorwave)
     carrierfrequency += 2
     modulatorfrequency -= 7
     duration = irand(2, 5) * 3.14
     }

//Day 12
rtsetparams(44100, 2)
load("WAVETABLE")
randomtest = irand(0, 100)
duration = 60
amplitude = 5000
frequency = cpspch(7.11)
waveform = maketable("wave", "nonorm", "nointerp", 2000, "sine")
pan = makeLFO("sine", "nointerp", 5.0, 0,1)

increment = 1.0
for(start = 0; start < 120; start += increment){
     if(randomtest < 50){
          duration = 5
          }
     else{
          duration = 10
          }
     WAVETABLE(start, duration, amplitude, frequency,        pan, waveform)
     WAVETABLE(start, duration, amplitude, frequency - 10.5, pan, waveform)
     }

//Day 13
rtsetparams(44100, 2)
load("WAVETABLE")
load("ELL")
```

```
bus_config("WAVETABLE", "aux 0-1 out")
bus_config("ELL", "aux 0-1 in", "out 0-1")
amplitude = 5000
wavetype = maketable("line", 32767, 0,0, 16384,1, 16385,-1, 16386,0, 32767,0)
pbcut = 1000      // passband cutoff frequency in Hz
sbcut = 90     // stopband cutoff frequency in Hz
ripple = 0.2      // amount of ripple (dB)
attenuation = 90  // attenuation at stopband (dB)
ellamp = 9        // filter amplitude
ringdur = .8      // ring-down duration (in sec.)
for(start = 0; start < 80; start += irand(0, 1)){
     frequency = irand(0, 1.0) / 7
     duration = irand (0, 1)
     pan = makeLFO("sine", 0.5, 0, 1)
     pbcut += irand(100, 400)
     WAVETABLE(start, duration, amplitude, frequency, 0.5, wavetype)
     WAVETABLE(start+1, duration+frequency, amplitude, frequency, 0.5, wavetype)
     ellset(pbcut, sbcut, 0, ripple, attenuation)
     // passband value > stopband = HPfilter
     ELL(start, 0, duration, ellamp, ringdur, 0, pan)
     }

//Day 14
rtsetparams(44100, 2)
load("FMINST")
duration = 0.7
amplitude = 5000
envelope = maketable("line", 1000, 0, 0, 3.5,1, 7,0)
carrier = cpspch(7.00)
modulatorfrequency = 220
minindex = 0
maxindex = 10
pan = maketable("random", 5, "gaussian", 0,1)
waveform = maketable("wave", 1000, "sine")
guide = maketable("line", "nonorm", 1000, 0, 0, 5,1, 7, 0)

increment = 0.25
for(start = 0; start < 100; start += increment){
     duration = irand(0.01, 0.8)
     FMINST(start, duration, amplitude*envelope, carrier, modulatorfrequency,
     minindex, maxindex, pan, waveform, guide)
     carrier = cpspch(irand(2.07, 8.09))
```

```
    guide = maketable("line", "nonorm", 1000, 0,1, 7,0)
    FMINST(start+1, duration*4, amplitude, carrier, modulatorfrequency, minindex,
    maxindex, pan,
    waveform, guide)
    maxindex = irand(10, 15)
    increment = irand(0, 1)
    }


//Day 15
rtsetparams(44100, 2)
load("WIGGLE")
amplitude = maketable("line", "nonorm", 1000, 0,0, 0.1,2000, 5,4000, 10,2000)
pitch = 2.03
envelope = maketable("curve", 2000, 0,0,2, 2.5,1,0, 13,1,-3, 25,0)
carrierwaveform = maketable("wave", 24051, "sine")
min = -1.00
max = 2.00
seed = srand()
gliss = maketable("random", "nonorm", "nointerp", 500, "low", min, max)
//p4 controls rate
freq = makeconverter(octpch(pitch) + gliss, "cpsoct")
mod_depth_type = 2 // 0 is no mod, 1 is % of carrier, 2 is FM
mod_wavetable = maketable("wave", 1000, "sine")
mod_freq = 200
mod_depth = 20
filt_cf = maketable("curve", "nonorm", 2000, 0,1000,-4, 1,1)
pan = maketable("line", "nonorm", 1000, 0,.2, 1,.5, 2,1)
filt_type = 0      // 0 is no filt, 1 is low, 2 is hight
filt_steep = 20
balance = true     // balance output and input signals

WIGGLE(start = 0, 60, amplitude * envelope, freq, mod_depth_type, filt_type, filt_steep,
balance,carrierwaveform, mod_wavetable, mod_freq, mod_depth, filt_cf, pan)

pan = maketable("line", "nonorm", 1000, 0,1, 1,0)
freq = makeconverter(octpch(pitch + 0.01) + gliss, “cpsoct”)

WIGGLE(start = 1.0, 60, amplitude * envelope, freq, mod_depth_type, filt_type,
filt_steep, balance, carrierwaveform, mod_wavetable, mod_freq, mod_depth, filt_cf, pan)
```

```
//Day 16
rtsetparams(44100, 2)
load(“JGRAN")
srand()

duration = 20
amplitude = 30.0
randomseed = srand()
oscconfig = 0 // wavetable, as opposed to FM (1)
oscphase = 1  // randomize osc phase? 0 is no; 1 is yes
grainenv = maketable("window", 1000, "hamming")
grainwaveclickhi = maketable("line", 32768, 0,0, 16384,0, 16385,1.0, 16387,-1.0,
16388,0, 32768,0)
grainwaveclicklow = maketable("line", 32768, 0,0, 15384,0, 16385,1.0,
17387,-1.0, 18388,0, 32768,0)
FMmult = maketable("random", 1000, "gaussian", 2.0, 100.0)
FMindex = 3.0
minfreq = 0.001 // really slow things down here to produce even faintest freq.
maxfreq = 12.0
minspeed = maketable("line", "nonorm", 1000, 0,0.2, 1,0.001) // decreasing minimum
maxspeed = maketable("line", "nonorm", 1000, 0,0.001, 1,0.2) // increasing maximum
mindb = 20
maxdb = 80
density = maketable("random", 1000, "gaussian", 0,10)
pan = makeLFO("sine", 0.5, 0,1)
panrand = maketable("random", 1000, "gaussian", 0,1)

for (start = 0; start < 80; start += 1){
     JGRAN(start, duration, amplitude, randomseed, oscconfig, oscphase, grainenv,
     grainwaveclickhi,
     FMmult,FMindex, minfreq, maxfreq, minspeed, maxspeed, mindb, maxdb, density,
     pan, panrand)
     }

//Day 17
rtsetparams(44100, 2)
load("VOCODESYNTH")
rtinput(“/path/to/file.wav“)

start = 0
instart = 0.0
duration = DUR()
```

```
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, .1,1, 0.6,1, 1.0,0)
numbands = 25
lowcf = 300
interval = 0.025
cartransp = 0.00
bw = 0.009
winlen = 0.001
smooth = 0.98
thresh = 0.0001
atktime = 0.001
reltime = 0.01
hipassmod = 0.0
hipasscf = 2000
carwavetable = maketable("wave", 10, 20000, "sine")
scale1 = 0.5
scale2 = 1.0
scalecurve = maketable("curve", "nonorm", 100, 0,scale1,1, 1,scale2)
spacemult = cpspch(interval) / cpspch(0.0)

VOCODESYNTH(start, instart, duration, amplitude*envelope, numbands, lowcf, spacemult,
cartransp, bw, winlen,smooth, thresh, atktime, reltime, hipassmod, hipasscf, inchan=0,
pan=1, carwavetable, scalecurve)

cartransp += 0.001
spacemult += 0.002

VOCODESYNTH(start, instart, duration, amplitude*envelope, numbands, lowcf, spacemult,
cartransp, bw,winlen,smooth, thresh, atktime, reltime, hipassmod, hipasscf, inchan=0,
pan=0, carwavetable, scalecurve)

//Day 18
rtsetparams(44100, 2) //exploring 3n+1 problem from G.E.B
load("WAVETABLE")
srand()

envelope = maketable("line", 1000, 0,0, 0.1,1, 0.3,0.7, 0.7,0.7, 1.0,0)
n = trunc(irand(10, 100)) //choose random integer between 10 and 100

increment = 0.1515
for(start = 0; start < 10; start += increment){
     if(n%2 == 0){ // if n is even
```

```
     n = trunc(n * 3 + 1)  // do a 3n + 1 operation
     transp = 1.0  // and transpose up and octave
     amplitude = irand(3000, 7000) // louder amplitude values
     pan = irand(0.0, 0.5) //pan L
     }
else if (n%2 == 1){ // if n is odd
     n = trunc(n / 2)  // divide n by 2
     transp = -1.0  // and transpose down an octave
     amplitude = irand(500, 2000)  //quieter
     pan = irand(0.5, 1.0) //pan R
     }
if (n == 1){ // if n equals 1
     exit()  // immediately stop program
     }
pitch = cpsmidi(n) // translate n to pitch in MIDI
if(pitch > 128){
     pitch = pitch - 127
     }
 dB = dbamp(amp)
 pan = irand(0.0, 1.0)
 constpowpan = boost(pan) / 2
  WAVETABLE(start, increment*0.8, amplitude*envelope, pitch, constpowpan)
print(n) // only print current n values
}


//Day 19
rtsetparams(44100, 2)
load("WAVETABLE")
reset(4)
srand(11)

envelope1 = maketable("line", 1000, 0,0, 5,1, 10,0)
envelope2 = maketable("line", 1000, 0,1.0, 5,0.5, 10,0)
waveform1 = maketable("random", 1000, "gaussian", -1,1)
waveform2 = maketable("wave3", 1000, 1,1,0, 2,0.05,0, 3,0.03,0, 4,0.02,0, 5,0.02,0,
6,0.005,1, 7,0.005,0,8,0.001,0, 9,0.001,0, 10,0.0001,0, 11,0.0001,0, 13,000.2,0,
15,0.0005,1, 17,0.0001,0,19,0.0001,0, 21,0.00002,0, 23,0.0005,1, 25,0.0001,0,
27,0.0001,0, 29.2,0.0002,0,31.7,0.00005,1, 33,0.00001,0, 35.5,0.000001,0, 37,0.00005,1,
39,0.00001,0, 41,0.00001,0, 43,0.00001,0 )
gliss1 = maketable("line", "nonorm", 1000, 0,1, 1.0,0.5)
gliss2 = maketable("line", "nonorm", 1000, 0,0, 1.0,1.0)
```

```
amp1 = 1000
amp2 = 1500
amp3 = 2000
pan1 = makeLFO("square", 0.1, 0.1,0.5)
pan2 = makeLFO("sine",  0.25, 0.5,1.0)
pan3 = makeLFO("buzz",  0.5,  0.0,1.0)
for (start = 0; start < 50; start += 1){
     WAVETABLE(start+irand(0, 1),   10, amp3*envelope1, 60*gliss1, pan1, waveform1)
     WAVETABLE(start+irand(0.5, 2), 9,  amp2*envelope2, 120*gliss2, pan2, waveform2)
     WAVETABLE(start+irand(0.5, 3), 10, amp1*envelope1, 240*gliss1, pan3, waveform1)
     WAVETABLE(start+irand(0.5, 1), 9.5,amp1*envelope2, 480*gliss2, pan1, waveform2)
     WAVETABLE(start+irand(0.5, 3), 10, amp2*envelope1, 760*gliss1, pan2, waveform1)
     WAVETABLE(start+irand(0, 1),   10, amp3*envelope2, 24051*gliss2, pan3, waveform2)
     start += irand (1, 5)
     if(start <= 25){
          amp1 += 1000
          amp2 += 1000
          amp3 += 1000
          }
     }
```

```
//Day 20
rtsetparams(44100, 2)
load("MBANDEDWG")
srand()

duration = 0.5
amplitude = 10000
pitch = cpsmidi(70)
strikeposition = 0.3 //range 0.0-1.0
pluck = 0 // 0 no pluck, 1 pluck
maxvelocity = 0.5 //range 0.0-1.0
instrument = 1 // 0 uniform bar, 1 tuned bar, 2 glass harmonica, 3 tibetan singing bowl
pressure = 0.0 //0.0 if struck, otherwise bowed
resonance = 0.9 //range 0.0-1.0
constant = 0.8 //range 0.0-1.0
pan = makeLFO("sine", 0.2, 0.0,1.0)
n = 11700 //choose random integer between 10 and 100

increment = 0.125
for(start = 0; start < 100; start += increment){
```

```
makegen(1, 24, 1000, 0,1, 2,0) //old style line graph for envelope
MBANDEDWG(start, duration, amplitude, pitch, strikeposition, pluck, maxvelocity,
instrument, pressure, resonance, constant, pan)
if(n%2 == 1){ // if n is odd
     n = trunc(n * 3 + 1)  // do a 3n + 1 operation
     constant = irand(0.0, 1.0)
     resonance = irand(0.0, 1.0)
     amp = irand(3000, 7000) // louder amplitude values
     }
else if (n%2 == 0){ // if n is even
     n = trunc(n / 2)  // divide n by 2
     maxvelocity = irand(0.0, 1.0)
     strikeposition = irand(0.0, 1.0)
     amp = irand(500, 2000)  //quieter
     }
pitch = cpsmidi(n) // translate n to pitch in MIDI
if(pitch < 30){
     pitch = pitch + 127
     }
}
```

```
//Day 21
rtsetparams(44100, 2)
load("HALFWAVE")
rate = control_rate(16)
print_off()
srand()

start = 0
duration = 0.75
pitch = octpch(5.07) //start on C#, can also use Hz with HALFWAVE()
amplitude = 300
envelope = maketable("window", 1000, "hamming")
wave1 = maketable("wave3", 1000, 3.14,1,0, 6.28,1,0.5)
wave2 = maketable("wave3", 1000, 1.00,1,0, 2.00,1,0.5)
wave3 = maketable("wave3", 1000, 1,1,0, 3,0.3,0, 5,0.2,0, 7,0.05,0, 9,0.01,0, 11,0.001,0)
wave4 = maketable("wave3", 1000, 1,1,0, 2,0.5,0, 3,0.3,0, 4,0.25,0, 5,0.2,0, 6,0.16,0,
7,0.14,0, 8,0.125,0) wave5 = maketable("wave3", 1000, 1,1,0, 3,0.14,0, 5,0.04,0,
7,0.02,0, 9,0.012,0, 11,0.008,0)
wavegamut = {wave1, wave2, wave3, wave4, wave5}
wavegamutlength = len(wavegamut)
```

```
wavecrossoverpoint = irand(0,1)
pan = 1

loop = 1.0
for(start = 0; start < 500; start += loop) {
     waveindex = trunc(irand(0,wavegamutlength))
     wave = wavegamut[waveindex]
     wavecrossoverpoint = irand(0,1)
     HALFWAVE(start, duration, pitch, amplitude*envelope, wave, wave, wavecrossoverpoint,
     pan*irand(0,1))

     waveindex = trunc(irand(0,wavegamutlength))
     wave = wavegamut[waveindex]
     wavecrossoverpoint = irand(0,1)
     pitch += (irand(-0.01,0.001) + trunc(irand(-2, 2)))
     HALFWAVE(start+2, duration+1, pitch, amplitude*envelope, wave, wave,
     wavecrossoverpoint, makeLFO("sine", 17, 0,1))

     waveindex = trunc(irand(0,wavegamutlength))
     wave = wavegamut[waveindex]
     wavecrossoverpoint = irand(0,1) // 0-1
     pitch += (irand(-0.01,0.001) + trunc(irand(-2, 2)))

     HALFWAVE(start+3, duration+2, pitch, amplitude*envelope, wave, wave,
     wavecrossoverpoint, makeLFO("sine", 18, 0,1))
     loop = irand(0,5)
     pitch += (irand(-0.01,0.001) + trunc(irand(-2, 2)))
     duration = irand(0, 5)
     amplitude = irand(200, 700)
     rate = round(irand(4, 128))
     }

//Day 22
rtsetparams(44100, 2)
load("CLAR")  //early clarinet physical model
load("MCLAR") //another
load("MBLOWHOLE") //another, with tonehole and register vent
seed = srand()
//--------------------//CLAR( ) parameters. Only create envelopes with makegen( )
clarduration = 2      //Original physical model by Perry Cook
clarnoiseamp = 0.001
clarlength1 = 400     // sample length1 (0-500)
```

```
clarlength2 = 99     // sample length2 (0-500)
claramplitude = 1000// absolute in 16-bit (0-32768)
clargain = 0.9
// 0-1.0, p2 gain is determined by comb.ofsample lenghts/d2
clarpan = spray_init(0, 10, seed) //unrepeated random  numbers (table 0, 10 elements...)
//--------------------//MCLAR( ) parameters. Can use maketable( ) for envelope control
mclarduration = 2
mclaramplitude = 1051
mclarfrequency = makeLFO("sine", 0.5, pchcps(440),pchcps(9000))
mclarmaxpressure = 0.5
mclarreedstiffness = 1.0
mclarpan = spray_init(1, 10, seed)
mbreathpressure = maketable("random", 25, "gaussian", 0,1.0)
//--------------------//MBLOWHOLE( ) parameters. Adds tonehole/vent
mblowduration = 2
mblowamplitude = 1051
mblowfrequency = makeLFO("saw", 0.55, pchcps(110),pchcps(5000))
mblownoiseamp = 0.5
mblowmaxpressure = 0.75
mblowreedstiffness = 0.25
mblowtoneholestate = 0  //0 is closed, 1 is open
mblowventstate = 1      //same
mblowpan = spray_init(2, 10, seed)
mblowbreathpressure = maketable("random", 25, "low", 0,1.0)
for(start = 0; start < 25; start += 1){
     modifier = irand(0,1.0)
     panvalue = get_spray(0)
     pan = panvalue / 10
     CLAR(start, clarduration, clarnoiseamp, clarlength1, clarlength2, claramplitude,
     clargain*modifier, pan)
     panvalue = get_spray(1)
     pan = panvalue / 10
     MCLAR(start+2, mclarduration, mclaramplitude, mclarfrequency, mclarmaxpressure,
     mclarreedstiffness,pan, mbreathpressure*modifier)
     panvalue = get_spray(2)
     pan = panvalue / 10
     MBLOWHOLE(start+3, mblowduration, mblowamplitude, mclarfrequency, mblownoiseamp,
     mblowmaxpressure,mblowreedstiffness, mblowtoneholestate, pan,
     mblowbreathpressure*modifier)
     }
```

```
//Day 23
rtsetparams(44100, 2) // Risset bell demo, after Charles Dodge 2nd ed, p 105  -JG
load("WAVETABLE")     // from John Gibon's RTcmix tutorial example, adapted JS
seed = srand()
control_rate(4)
env1 = maketable("expbrk", 1000,  1, 1000, .0000000002) // more attack
env2 = maketable("line", 1000, 0,0, 0.3,0.3, 0.6,0.3, 0.75,0.7, 1.0,0.0) //reverse ADSR
wavet = maketable("wave", 100, "sine");
durscale = 3
notes = {
//  start  dur   amp   freq
     {  0,   52,  3000,  82051 },
     {  2,   40,  4000,  91200 },
     {  5,    5,  2000,  74800 },
     {  7,   34,  1500,     50 },
     {  8,   26,  1000,  62576 },
     { 10,   11,  2000,  82709 },
     { 11,   17,  2000,  95934 },
     { 13,   20,  4000,   1348 },
     { 14,   18,  1000,     60 },
     { 16,   22,  3000,   1102 },
     { 17,   28,  1500,  55626 },
     { 20,    7,  2500,    895 },
     { 22,    8,  2500,    880 },
     { 26,   16,  3500,    220 },
     { 28,   10,  2000,    110 },
     { 34,   13,  2000,     55 },
     { 40,   14,  1050,    110 },
     { 52,    2,  1000,    220 }
     }
numnotes = len(notes)  // set up an array for the bell notes
paninit = spray_init(0, 10, seed)

loop = 1.0
for (i = 0; i < numnotes; i += 1) {
     note = notes[i]
     start = note[0]
     dur = note[1] * durscale
     amp = note[2] * env1
     freq = note[3]
     panvalue = get_spray(0)
     pan = panvalue / 10    //random panning between 0 and 1.0
```

```
    WAVETABLE(start, dur,      amp,      freq*.56,     pan, wavet)
    WAVETABLE(start, dur*.9,   amp*.67,  freq*.56+1,   pan, wavet)
    WAVETABLE(start, dur*.65,  amp,      freq*.92,     pan, wavet)
    WAVETABLE(start, dur*.55,  amp*1.8,  freq*.92+1.7, pan, wavet)
    WAVETABLE(start, dur*.325, amp*2.67, freq*1.19,    pan, wavet)
    WAVETABLE(start, dur*.35,  amp*1.67, freq*1.7,     pan, wavet)
    WAVETABLE(start, dur*.25,  amp*1.46, freq*2,       pan, wavet)
    WAVETABLE(start, dur*.2,   amp*1.33, freq*2.74,    pan, wavet)
    WAVETABLE(start, dur*.15,  amp*1.33, freq*3,       pan, wavet)
    WAVETABLE(start, dur*.1,   amp,      freq*3.76,    pan, wavet)
    WAVETABLE(start, dur*.075, amp*1.33, freq*4.07,    pan, wavet)
    loop = pan
    }


//Day 24
rtsetparams(44100, 2)
load("MMODALBAR")
load("DELAY")
load("IIR")
print_off()
reset(8)
srand()

bus_config("MMODALBAR", "aux 0-1 out")
bus_config("DELAY", "aux 0-1 in", "out 0-1")
loop = 0.125

//-------------------Modal Bar parameters
barstart = 0
barduration = 3.0
baramplitude = 20000
barnote = 44051
stickhardness = 1.0  //0.0-1.0
stickposition = 0.1  //0.0-1.0
barinstrument = 1
//wood...0marimba, 1vibe, 2agogo, 3wood1, 4reso, 5wood2, 6beats, 7twofixed, 8clump
barpan = makeLFO("sine", 20.2, 0,1)

//-------------------Delay parameters
delayinstart = 0
totalduration = 2
```

```
delayamplitude = 0.8
delaytime = 1.0 //seconds
delayfeedback = 0.8
ringdownduration = 1.0
inputchannel = 0
for(start = 0; start < 300; start += loop){
     makegen(1, 24, 1000, 0,1, 1, 1) // old school envelope
     MMODALBAR(start, barduration, baramplitude, barnote, stickhardness, stickposition,
     barinstrument,barpan)
     barnote -= irand(10, 30)

     MMODALBAR(start+2, barduration, baramplitude, barnote, stickhardness, stickposition,
     barinstrument,barpan-0.5)
     delaypan = random()

     DELAY(start, delayinstart, totalduration, delayamplitude, delaytime, delayfeedback,
     ringdownduration, inputchannel, delaypan)
     loop += 0.015
     }

filteroutstart = 0
filterinstart = 1
filterduration = 40
filteramplitude = 20000
filterenvelope = maketable("window", 1000, "hamming")
filterpitch = 44
filterpan = 0.5
center1 = 300

for(i = 0; i < 100; i += 1.0){
     setup(center1,25.0,0.8, 1200.0,15.0,0.9) //CF1,BW1,AMP1, CF2,BW2,AMP2...
     PULSE(i, filterinstart, filterduration, filteramplitude*filterenvelope, filterpitch,
     filterpan, filterpitch += irand(10.0, 40.0)
     center1 += 100.0
     }


//Day 25
rtsetparams(44100, 2, 16)
load("MULTIWAVE")   //Additive synthesis instrument
control_rate(32)    //Adapted from JGs 3/10/05 help file
```

```
duration = 180
masteramp = 10000
minfreq = 220
maxfreq = 6000
glide = 20
quantum = 100 // quantize freqs to this number (in Hz)
waveform = maketable("wave3", 1000, 1,1,0, 3,0.14,0, 5,0.04,0, 7,0.02,0, 9,0.012,0,
11,0.008,0)
envelope = maketable("line", 1000, 0,0, 0.2,0.4, 0.6,0.4, 0.7,1, 1.0,0)
numwaves = 12
freq = {}  // freq index
pan = {}    // pan index

loop = 1
for (i = 0; i < numwaves; i += loop) {
     masteramp = irand(15000, 20000)
     lfofreq = 0.007 + (i * 1.4)
     rfreq = makeLFO("sine", lfofreq, min = 0.2 + (i * 0.03), min * 3.5)
     min = minfreq + (i * 10)
     max = maxfreq - (i * 70)
      rand = makerandom("linear", rfreq, min, max, seed = i + 1)
     freq[i] = makefilter(rand, "smooth", glide)
     if (quantum){
               freq[i] = makefilter(freq[i], "quantize", quantum)
         }
     min = mod(i, 2)

     if (min == 0){
               max = 1
               }
     else{
          max = 0
          }

     pan[i] = makeLFO("sawup", 0.007 + (i * 0.026), min, max)
     loop = irand(0,1)
     }

amp = makerandom("cauchy", 100, 0,1)
phase = makerandom("high", 100, 0,1)

MULTIWAVE(start = 0, duration, masteramp*envelope, waveform,
```

```
   freq[0], amp, phase, pan[0],
   freq[1], amp, phase, pan[1],
   freq[2], amp, phase, pan[2],
   freq[3], amp, phase, pan[3],
   freq[4], amp, phase, pan[4],
   freq[5], amp, phase, pan[5],
   freq[6], amp, phase, pan[6],
   freq[7], amp, phase, pan[7],
   freq[8], amp, phase, pan[8],
   freq[9], amp, phase, pan[9],
   freq[10],amp, phase, pan[10],
   freq[11],amp, phase, pan[11])


//Day 26
rtsetparams(44100, 2, 512)  //Sound vaguely like really bad Nintendo
load("WAVETABLE")
load("FLANGE")
control = 128
control_rate(control)
srand()
print_off()

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("FLANGE", "aux 0-1 in", "out 0-1")

envelope1 = maketable("line", 1000, 0,0, 5,1, 10,0)
envelope2 = maketable("line", 1000, 0,1.0, 5,0.5, 10,0)
waveform1 = maketable("random", 5, "gaussian", -1,1)
waveform2 = maketable("wave", 1000, "sawdown", 0,0, 1,0, 2,1, 3,1)
glissup =   maketable("line", 1000, 0,0.1, 1.0,0)
glissdown = maketable("line", 1000, 0,0, 1.0,1)
pan1 = makeLFO("square",0.1, 0.1,0.5)
pan2 = makeLFO("sine",  5.25,0.5,1.0)
pan3 = makeLFO("buzz",  10.5, 0.0,1.0)
amp1 = 1000
amp2 = 1500
amp3 = 2000
for (st = 0; st < 1000; st += 1){
     WAVETABLE(st+irand(0, 1),   1, amp3*envelope1, .3, pan1, waveform2)
     WAVETABLE(st+irand(0.5, 2), 5, amp2*envelope2, 3, pan2, waveform2)
     WAVETABLE(st+irand(0.5, 3), 3, amp1*envelope1, .24*glissup, pan3, waveform2)
```

```
WAVETABLE(st+irand(0.5, 1), 9, amp1*envelope2, 8*glissdown, pan1, waveform2)
WAVETABLE(st+irand(0.5, 3), 4, amp2*envelope1, .7, pan2, waveform1)
WAVETABLE(st+irand(0, 1),   2, amp3*envelope2, 5*glissup, pan3, waveform2)

     if(st%2 == 0){
          amp1 += 100
          amp2 += 100
          amp3 += 100
          }

     if(st%2 == 1){
          amp1 -= 100
          amp2 -= 100
          amp3 -= 100
          }

flangedur = irand(0, 5)
resonance = irand(0,1)
lowpitch = irand(0,4)
moddepth = irand(0,5)
modspeed = irand(0,1)
     wetdrymix = irand(0,1)
     maxdelay = 1.0 / cpspch(lowpitch)

     FLANGE(st, 0, flangedur, 0.8, resonance, maxdelay, moddepth, modspeed, wetdrymix,
"IIR", 0, pan=1,ringdur=0, waveform1)
     FLANGE(st, 0, flangedur*0.75, 0.8, resonance, maxdelay, moddepth, modspeed, wet
drymix, "IIR", 0, pan=0, ringdur=0, waveform2)

      control = pickrand(2, 4, 8, 16, 32, 64, 128, 256, 512)
      control_rate(control)
      loop = pickrand(2.5, 3, 1.0, 5.0, 0.1)
      }

//Day 27
rtsetparams(44100, 2, 128)
load("WAVETABLE")
load("MMODALBAR")
control = 44100
reset(control)
srand()
print_off()
```

```
//--------------cascading waveforms (weird, random, jaggedy looking things
st = 0
dur = 10
amp = 1000
env1 = maketable("line", 1000, 0,0, 0.1,1, 0.9,1, 1.0,0)
env2 = maketable("line", 1000, 0,0.8, 1,0.0)
freq = 22051
pan1 = makeLFO("sine", 60.5, 0,0.5)
pan2 = makeLFO("sine", 50.75, 0.5,1)
waveform = maketable("random", 8, "gaussian", -1,1)
loop = 1

for(st = 0; st < 125; st += loop){
     dur = irand(5, 20)
     WAVETABLE(st,   dur, amp*env1, freq,   pan1, waveform)
     freq -= 100
     WAVETABLE(st+2, dur/2, amp-200*env2, freq/2, pan2, waveform)
     freq += 10
     WAVETABLE(st+3, dur/3, amp-400*env1, freq/3, pan1, waveform)
     freq -= 5
     WAVETABLE(st+4, dur/4,  amp-600*env2, freq/4, pan2, waveform)
     freq += 2.5
     WAVETABLE(st+5, dur/5, amp-800*env1, freq/5, pan1, waveform)
     freq -= 100
     WAVETABLE(st+6, dur/6,  amp*env2,  freq/6, pan2, waveform)
     loop = irand(0, 1)
     control = pickrand(16, 32, 128, 256, 4096, 22050, 44100)
     control_rate(control)
     }
//---------------sort of a John Adams' Short Ride simple repeating block pattern
makegen(1, 24, 1000, 0,1, 1, 1)
start = 0
duration = 1.0
amplitude = 20000
frequency = 22051
hardness = 0.0
position = 0.4
instrument = 0 // 0marimba, 1vibe, 2agogo, 3wood1, 4reso, 5wood2, 6beats, 7fixed, 8clump
pan1 = makeLFO("square", 0.25, 0,0.6)
pan2 = makeLFO("sine", 0.3333, 0.4,1)

for (i = 0; i < 25; i +=1){
```

```
	hardness = 0.0
	for (ii = 0; ii < 50; ii += 1){
		MMODALBAR(start, duration, amplitude, frequency, hardness, position, i, pan1)
		hardness += 0.01
		start += 0.5
		frequency -= 0.5
		MMODALBAR(start*(ii/2), duration, amplitude, frequency, hardness, position, i,
pan1)
		}
	MMODALBAR(start, duration, amplitude, frequency, hardness-0.4, position, i*2, pan2)
	frequency += 1.33
	MMODALBAR(start*(i/2), duration, amplitude, frequency, hardness-0.4, position, i,
pan2)
	}


//Day 28
rtsetparams(44100, 2)
load("COMPLIMIT")
load("MOCKBEND")
load("PANECHO")
rtinput("/path/to/file.aiff")

bus_config("COMPLIMIT", "in 0-1", "aux 0-1 out") //needs to read in from 0-1?
bus_config("MOCKBEND", "aux 0-1 in", "aux 2-3 out")
bus_config("PANECHO", "aux 2-3 in", "out 0-1")

//-----------------compression stuff
start = 0
instart = 0
dur = DUR()
ingain = 18
outgain = 0
attack = 0.001
release = 0.02
threshold = -20
ratio = 4
lookahead = attack
windowlen = 128
detect_type = 0
bypass = 0
inchan = 0
pan = 0.5 //no further panning, input file is already interesting enough
COMPLIMIT(start, instart, dur, ingain, outgain, attack, release, threshold, ratio,
```

```
lookahead, windowlen, detect_type, bypass, inchan, pan)

//------------------MOCKBEND, uses old-style makegens
start = 0
instart = 0
dur = DUR()
amp = 0.7
pan = 0.5
setline(0,0, 1, 1, 90, 1, 100, 0) // amplitude curve
makegen(-2, 18, 512, 1,.5, 512,-.11) //transpose up 5 semitones, then down 11
MOCKBEND(start, instart, dur, amp, 2, 0, pan)

//------------------ping-pong delay
start = 0
instart = 0
dur = DUR()
amp = 0.7
channel0delay = 2.12  // in sec
channel1delay = 0.55
regenerator = 0.8     // multiplier, ALWAYS less than 1!
ringdowndur = 7.2
makegen(1, 24, 1000, 0,1, 100, 1)
PANECHO(start,    instart, dur,   amp, channel0delay, channel1delay, regenerator,
ringdowndur)

for(i = 0; i < 25; i += 1){
     channel0delay += irand(0,1)
     channel1delay = irand(0,1)
     regenerator = irand(0,1) - 0.1
     }

PANECHO(start, instart, dur-4, amp, channel0delay, channel1delay, regenerator,
ringdowndur-1.2)



//Day 29
rtsetparams(44100, 2)    // from JG 6/3/2002
load("MIX")
load("VOCODE2")
rtinput(“/path/to/file.aiff”)

//------------- carrier for vocoder
```

```
bus_config("MIX", "in 0", "aux 0 out") // separate between channels
instart = 0
amp = 0.8
dur = DUR() - instart
MIX(0, instart, dur, amp, 0)
//------------- modulator for vocoder
bus_config("MIX", "in 0", "aux 1 out")
instart = 0
dur = DUR() - instart
amp = 0.8
MIX(0, instart, dur, amp, 0)
// ------------- vocoder
bus_config("VOCODE2", "aux 0-1 in", "out 0-1")
//list of center frequencies
cftabs = maketable("literal", "nonorm", 0, 3.00, 4.07, 5.06, 6.02, 7.01, 7.04, 7.11,
8.06, 9.01, 9.03)
numpitches = tablelen(cftabs)
outstart = 0
instart = 0
dur = DUR() + 2  // allow vocoder to ring, if nec.
maxamp = 1.0
amplitude = maketable("line", "nonorm", 1000, 0,maxamp, dur-1,maxamp, dur,0)
numfilt = 0   // number of filters. this is a flag, make sure its 0
transp = 0.2
freqmult = 4.02
cartransp = -0.02
bw = 0.008
resp = 0.0001

for(ii = 0; ii < 10; ii += 1){
     resp = random() / 10
     }

hipass = 0.1
hpcf = 5000
noise = 0.01
noisubsamp = 4

for(i = 0; i < 25; i += 1){
     transp += 0.01
     freqmult = random() + 3
     amp = add(amplitude, irand(0,1))   // add constant to table!
```

```
     cftab = modtable(cftabs, "shift", round(irand(1,4))) // shift contents

     VOCODE2(i, instart, 1, amp, numfilt, transp, freqmult, cartransp, bw, resp, hipass,
     hpcf, noise, noisubsamp, 0.5, cftab)
      }


//Day 30
rtsetparams(44100, 2)
load("WAVETABLE")
load("JCHOR")
load("SPECTACLE2")
reset(22050)
print_off()
srand()

rtinput("/Users/jerod_s/Desktop/bwv20.aif")

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("JCHOR", "in 0-1", "aux 0-1 out")
bus_config("SPECTACLE2", "aux 0-1 in", "out 0-1")
//------------------Wavetables
start = 0
dur = DUR()
amplitude = 1000.0
frequency = maketable("random", 250, "even", 250,2000)
waveform = maketable("wave", "nonorm", "nointerp", 1000, "buzz")
panfreq = maketable("random", 10, "gaussian", 0,1)
pan = makeLFO("sine", panfreq, 0,1)
WAVETABLE(start, dur, amplitude, frequency, pan, waveform)
WAVETABLE(start, dur, amplitude, frequency - 10.5, 0.5, waveform)
//------------------Chorus
outskip = 0
outdur = 9
instart = 0.00
indur = 0.20
maintain_dur = 1
transposition = -0.05
nvoices = 10
minamp = 0.1
maxamp = 0.5
```

```
minwait = 0.00
maxwait = 0.60
seed = srand()
setline(0,0, .5,1, outdur/8,1, outdur,0)
makegen(2, 7, 1000, 0, 10, 1, 990, 0)
JCHOR(outskip, instart, outdur, indur, maintain_dur, transposition, nvoices, minamp,
maxamp, minwait, maxwait, seed)
//------------------Spectacle
inchan = 0
instart = 0
ringdur = 15
amp = 1.0
wet = 0.8
fftlen = 1024
winlen = fftlen * 2
overlap = 2
window = 0
ienv = maketable("line", 1000, 0,0, 1,1, 19,1, 20,0)// input envelope
oenv = maketable("curve", 1000, 0,1,0, 2,1,-1, 3,0)// output envelope
mineqfreq = 0
maxeqfreq = 0
eqtablen = 1000
eq = maketable("line", "nonorm", eqtablen, 0,-90, 200,0, 8000,-3, 22050,-6, 44100, 0)
deltablen = fftlen / 2
mindelfreq = 0
maxdelfreq = 0
mindt = .4  // Delay times
maxdt = 3
seed = srand()
deltime = maketable("random", "nonorm", deltablen, "even", mindt, maxdt, seed)
minfb = .1  // Feedback times
maxfb = .8
fbtime = maketable("random", "nonorm", deltablen, "even", minfb, maxfb, seed)
print_on()
panfrequency = maketable("random", 10, "gaussian", 0,1)
print(panfrequency)
pan = makeLFO("saw", panfrequency, 0, 1)  // sine for smooth, saw for clicks (later on)
SPECTACLE2(start, instart, dur, amp*oenv, ienv, ringdur, fftlen, winlen, window, overlap,
eq, deltime,fbtime, mineqfreq, maxeqfreq, mindelfreq, maxdelfreq, 0, wet, inchan, pan)
```

```
//Day 31
rtsetparams(44100, 2)
control_rate(44100)
load("DCBLOCK")
load("TRANSBEND")
load("FIR")
load("DEL1")
load("REVERBIT")
load("WAVETABLE")
rtinput(“/path/to/file.aiff”)

bus_config("DCBLOCK", "in 0-1", "aux 0-1 out")
bus_config("TRANSBEND", "aux 0-1 in", "aux 2-3 out")
bus_config("FIR", "aux 2-3 in", "aux 4-5 out")
bus_config("DEL1", "aux 4-5 in", "out 0-1”)

//----------------DCBLOCK
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
DCBLOCK(start, instart, duration, amplitude)
//----------------TRANSBEND
start = 0
instart = 0
duration = DUR() // length of soundfile
amplitude = 0.8
inchan = 0
pan = 0.5
setline(0,0, 1,1, 90,1, 100,0) // draws straight line segments into array
makegen(0, 18, 512, 1,0.3, 512,-0.6) // transpose from three semitones up, to six down
TRANSBEND(start, instart, duration, amplitude, inchan, pan)
//----------------FILTER
start = 0
instart = 0
duration = DUR()
amplitude = 0.8
//remaining calculations are filter coefficients
FIR(start, instart, duration, amplitude, 7,0.9,0.1,0.69,-0.49,0.314,0.2,0.09)
//----------------DELAY
start = 0
instart = 0
```

```
duration = 60
amplitude = maketable("spline", 1000, "closed", curvature = 25, 0,0.1, 0.5,0.9,1.0,0)
delaytime = 3.14 // R channel delay time, L is default
rightamp = 1.0   // amplitude of R relative to L
DEL1(start, instart, duration, amplitude, delaytime, rightamp)
//--------------REV
start = 0
instart = 0
duration = 60
amplitude = 0.8
revtime = maketable("line", 1000, 0,0.01, 0.5,0.1, 1.0,0.2) //keep these short
revamnt = 1 //0 is dry, 1 is wet
chandelay = 0.01 //R channel delay to L channel
cutoff = 1000 //LOP~ cutoff in Hz
REVERBIT(start, instart, duration, amplitude, revtime, revamnt, chandelay, cutoff)
//--------------WAVETABLE
st = 0
dur = 10
amp = 1000
env = maketable("line", 1000, 0,0, 0.1,1, 0.9,1, 1.0,0)
freq = 22051
pan1 = makeLFO("sine", 60.5, 0,0.5)
pan2 = makeLFO("sine", 50.75, 0.5,1)
waveform = maketable("random", 8, "gaussian", -1,1)
loop = 0.25
for(st = 0; st < 50; st += loop){
     dur = irand(5, 20)
     WAVETABLE(st,   dur, amp*env, freq,  pan1, waveform)
     freq -= 100
     WAVETABLE(st+2, dur/2, amp-250*env,  freq/2, pan2, waveform)
     freq += 10
     loop = irand(0, 1)
     }


//Day 32
rtsetparams(44100, 2)
load("WAVETABLE")
load("PANECHO")
load("PAN")
control_rate(44100) // how often to update values in the score
srand() // seeded random value generator, empty? seed to clock time
```

```
bus_config("WAVETABLE", "aux 0-1 out")
bus_config("PANECHO", "aux 0-1 in", "aux 2-3 out")
bus_config("PAN", "aux 2-3 in", "out 0-1")

waveform1 = maketable("wave3", 10, 0,1, 1,1, 2,1, 3,0, 4,0.5, 5,1)
waveform2 = maketable("wave3", 1000, 0,1, 1,1, 2,1, 3,0, 4,0.5, 5,1)
waveform3 = maketable("wave", 1000, "sine")
waveform4 = maketable("wave", 1000, "saw99")
waveform5 = maketable("wave", 1000, "tri")
waveform6 = maketable("random", 20, "even", -1,1)
waveform7 = maketable("wave", 1000, "buzz")
waveform8 = maketable("wave", 10, "saw")  //"bit crushed" waves
waveform9 = maketable("wave", 10, "sine")

wavearray = {waveform1, waveform2, waveform3, waveform4, waveform5, waveform6, waveform7,
waveform8, waveform9}
wavelength = len(wavearray)

start = 0
duration = 14
amplitude = 20000  // typically between 1000 - 40000
envelope1 = maketable("line", 1000, 0,0, 0.3,1, 0.5, 0.7, 0.8,0.7, 1.0,0) //ADSR
envelope2 = maketable("random", 10, "gaussian", 0,1)
frequency = irand(10,2000)
pan = 0.5

for(start = 0; start < 1000; start += 1){

     if(start < 700){
          frequency = pickrand(7.00, 7.04, 7.07, 7.09, 8.00, 8.04, 8.07, 8.09) // A-7
          pan = makeLFO("sine", 0.0125, 0,1)
          WAVETABLE(start, duration, 1000*envelope1, frequency, pan, waveform1)
     }

     if(start >=250 && start <=700){
          waveindex = trunc(irand(0, wavelength))
          wave = wavearray[waveindex]
          frequency = pickrand(7.00, 7.04, 7.05, 7.09, 8.00, 8.04, 8.05, 8.09) //F maj 7
          amplitude = pickrand(100, 200, 250)
          pan = makeLFO("saw", 2, 0,1)
          duration = pickrand(1, 1, 2, 3, 5, 8, 13)
```

```
        WAVETABLE(start, duration, amplitude*envelope2, frequency, pan, wave)
    }

    if(start >=500){
        waveindex = trunc(irand(0, wavelength))
        wave = wavearray[waveindex]
        frequency = pickrand(6.10, 7.00, 7.02, 7.05, 7.07) // Bb pentatonic
        amplitude = pickrand(50, 75, 100)
        duration = irand(1,10)
        WAVETABLE(start, duration, amplitude*envelope1, frequency, pan, wave)
        }

    }

//------------------ping pong delay
start = 0
instart = 0
duration = 9*60 // three minutes long
amplitude = 0.8
envelope = maketable("line", 1000, 0,0, 0.5,1, 3.5,1, 7,0)
delaytimeleft = 3.14
delaytimeright = 1.07
feedback = 0.7
ringdownduration = 3
PANECHO(start, instart, duration, amplitude*envelope, delaytimeleft, delaytimeright,
feedback, ringdownduration)

//--------------------panning
start = 0
instart = 0
duration = 9*60
amplitude = 2.0
envelope = maketable("line", 1000, 0,0, 1,1, duration-1,1, duration,0)
inchannel = 0
panmode = 1        //0 for constant power pan, 1 for linear pan
dynamicpan = maketable("random", 20, "gaussian", -1,1)
paninvert = makefilter(dynamicpan, "invert", 0.5)
pan = makefilter(paninvert, "fitrange", -1.0,1.0)
PAN(start, instart, duration, amplitude*envelope, inchannel, panmode, pan)
```

```
//Day 33
rtsetparams(44100, 2)
load("METAFLUTE")     //flute physical model family of instruments
print_off()

//----------------------SFLUTE (most basic flute model)
start = 0
duration = 5
noiseamp = 0.01  //noise amplitude relative to overall
length1 = 20    //samples, between 5-200
length2 = 200
amplitude = 7000
pan = 0.5
/*
still uses the old style makegens for envelopes, one for noise amplitude and
another for the overall amplitude
*/
table = 1
gentype = 24 //
size = 1000
makegen(table, gentype, size, 0,1, 1.5,1)
table = 2
gentype = 24
size = 1000
makegen(table, gentype, size, 0,0, 0.05,1, 1.49,1, 1.5,0)

SFLUTE(start, duration, noiseamp, length1, length2, amplitude, pan)

//----------------------VSFLUTE (with vibrato)
start = 3
duration = 5
noiseamp = 0.1
length1lowvalue = 70     //create low and high values for vibrato depth
length1highvalue = 72
length2lowvalue = 40
length2highvalue = 43
amplitude = 7000
vibrato1freqlow =  0.5     //Hz
vibrato1freqhigh = 4.0
vibrato2freqlow =  1.9
vibrato2freqhigh = 3.2
```

```
pan = 0.5
/*
requires makegens
1) table 1 for noise amplitude envelope
2) table 2 for overall amplitude envelope
3) table 3 to build a waveform for length 1s frequencies
4) table 4 ditto, but for length 2
*/
table = 1
gentype = 7
size = 1000
makegen(table, gentype, size, 1, 1000, 1)

table = 2
gentype = 7
size = 1000
makegen(table, gentype, size, 1, 1000, 1)

table = 3
gentype = 10
size = 1000
makegen(table, gentype, size, 1)

table = 4
gentype = 10
size = 1000
makegen(table, gentype, size, 1)

VSFLUTE(start, duration, noiseamp, length1lowvalue,length1highvalue,
length2lowvalue,length2highvalue, amplitude, vibrato1freqlow,vibrato1freqhigh,
vibrato2freqlow,vibrato2freqhigh, pan)

//---------------------BSFLUTE (with pitch bending)
start = 7.0
duration = 5
noiseamp = 0.1
length1lowvalue = 90
length1highvalue = 100
length2lowvalue = 140
length2highvalue = 150
amplitude = 5000
pan = 0.5
```

```
//makegens are the same as for vibrato
table = 1
gentype = 7
size = 1000
makegen(table, gentype, size, 1, 1000, 1)

table = 2
gentype = 7
size = 1000
makegen(table, gentype, size, 1, 1000, 1)

table = 3
gentype = 10
size = 1000
makegen(table, gentype, size, 1)

table = 4
gentype = 10
size = 1000
makegen(table, gentype, size, 1)
```

```
BSFLUTE(start, duration, noiseamp, length1lowvalue,length1highvalue,
length2lowvalue,length2highvalue,amplitude, pan)


//Day 34
rtsetparams(44100, 2)
load("WAVETABLE")
load("MMODALBAR")
load("REV")
controlrate = control_rate(44100)
srand()
print_off()

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("MMODALBAR", "aux 0-1 out")
bus_config("REV", "aux 0-1 in", "out 0-1")

//---------------------------start with an array of control rates
rate_array = {2, 4, 8, 16, 32, 64, 128, 256, 512, 1024, 2048, 4096}
rate_array_size = len(rate_array)
random_rate = trunc(trand(0, rate_array_size))
```

```
//current_rate = rate_array[random_rate]

//---------------------------now to get a few pitch arrays
Cmaj7_array =  {7.00, 7.04, 7.07, 7.11}
Gmaj7_array =  {7.02, 7.06, 7.07, 7.11}  //put in second inversion to fit within one oc-
tave
Amin7_array =  {7.00, 7.04, 7.07, 7.09}  //first inversion
Bbmaj7_array = {7.02, 7.05, 7.09, 7.10}  // "

//---------------------------a few universals for the above
pitch_array = Cmaj7_array     //or any of the others...
pitch_array_size = len(pitch_array)
random_pitch = trunc(trand(0, pitch_array_size))
current_pitch = pitch_array[random_pitch]

//---------------------------on to the wavetables
start = 0
duration = 10
amplitude = 2000
//envelope = maketable("line", 1000, 0,0, 0.5,1.0, 1.0,0)
pitch_array = Cmaj7_array
pan = 0.5
waveform = maketable("wave3", 100, 0,1,0, 1,1,0, 2,0.5,0, 3,0.5,0, 5,1.0,0, 8,0.5,0,
13,0.25,0, 21,0.001,0)

//---------------------------for modal bars
barstart = 0
barduration = 3.0
baramplitude = 80000
barnote = cpspch(8.09)
stickhardness = 1.0  //0.0-1.0
stickposition = 0.1  //0.0-1.0
barinstrument = 3
barpan = makeLFO("sine", 0.2, 0,1)
loop = 2.5

for(start = 0; start < 200; start += 2.5){
     random_pitch = trunc(trand(0, pitch_array_size))
     current_pitch = pitch_array[random_pitch]
     pitch = current_pitch

     WAVETABLE(start, duration, amplitude, cpspch(pitch), pan, waveform)
     current_pitch = pitch_array[random_pitch]
```

```
pitch = current_pitch

WAVETABLE(start+1, duration, amplitude, cpspch(pitch)+1, pan, waveform)

barpan = random()
MMODALBAR(start, barduration, baramplitude, barnote, stickhardness, stickposition,
barinstrument,barpan)

if(start >= 75 && start <= 110){
     pitch_array = Gmaj7_array
     WAVETABLE(start, duration, amplitude, cpspch(pitch), pan, waveform)
     MMODALBAR(start+0.25, barduration, baramplitude, barnote, stickhardness,
     stickposition, barinstrument, barpan)
     }

if(start >= 90 && start <= 150){
     pitch_array = Amin7_array
     WAVETABLE(start, duration, amplitude, pitch, pan, waveform)
     }

if(start > 140){
          pitch_array = Bbmaj7_array
          WAVETABLE(start+0.9, duration, amplitude, pitch, pan, waveform)
          MMODALBAR(start, barduration, baramplitude, barnote, stickhardness,
          stickposition, barinstrument, barpan)
          }

     random_rate = trunc(trand(0, rate_array_size))
     current_rate = rate_array[random_rate]
     loop = random() + 2
     }

//---------------------------reverb
start = 0
instart = 0
duration = 60 * 5
amplitude = 0.9
type = 1     // 1 is Perry Cook's, 2 is John Chowning's, 3 is Michael McNabb's
rvbtime = 2.5
rvbpct = 0.5
inchan = 0
REV(start,   instart, duration, amplitude, type, rvbtime, rvbpct, inchan)
```

```
REV(start+1, instart, duration, amplitude, type, rvbtime, rvbpct, inchan)


//Day 34
rtsetparams(44100, 2)
load("WAVETABLE")
load("MMODALBAR")
print_off()

rate_array = {2, 4, 8, 16, 32, 64, 128, 44100}
rate_length = len(rate_array)
rate = rate_array[trand(0, rate_length)]

fibonacci_array = {1, 1, 2, 3, 5, 8, 13, 21, 34, 55, 89, 144, 233}
fibonacci_length = len(fibonacci_array)

//------------------------WAVETABLE
start = 0
duration = 1      //duration and amplitude taken care of by fibonacci array above
amplitude = 10000

env1 = maketable("line", 1000, 0,0, 0.4,1, 1,0.5, 10,0)
env2 = maketable("line", 1000, 0,0, 1,0.03, 2,0.1, 3,0.15, 4,0.2, 5,0.4, 6,0.6, 7,0.8,
7.5,1, 10,0)
env3 = maketable("line", 1000, 0,0, 1,0.1, 2,0.0, 3,0.4, 4,0.1, 5,0.6, 6,0.2, 7,0.9,
8,0.2, 9,1, 10,0)
env4 = maketable("line", 1000, 0,0, 5,1, 10,0)
env5 = maketable("line", 1000, 0,0, 1,0.1, 6,0.2, 9,1, 10,0)
env6 = maketable("line", 1000, 0,0, 0.4,1, 1,0.3, 2,0.1, 3,0.3, 4,0.1, 5,0.3, 6,0.1,
7,0.3, 8,0.1, 9,0.3, 10,0)
envelope_array = {env1, env2, env3, env4, env5, env6}
envelope_length = len(envelope_array)
envelope = envelope_array[trand(0, envelope_length)]

white_keys = pickrand(8.00, 8.02, 8.04, 8.05, 8.07, 8.09, 8.11)
black_keys = pickrand(8.01, 8.03, 8.06, 8.08, 8.10)
Cmaj7 = pickrand(7.00, 7.04, 7.07, 7.11, 8.00, 8.04, 8.07, 8.11)
Gmaj7 = pickrand(7.07, 7.11, 8.02, 8.06, 8.07, 8.11, 9.02, 9.06)
Amin7 = pickrand(7.09, 8.00, 8.04, 8.07, 8.09, 9.00, 9.04, 9.07)
Dmin7 = pickrand(7.02, 7.05, 7.09, 8.00, 8.02, 8.05, 8.09, 9.00)
Bmin7 = pickrand(7.11, 8.02, 8.06, 8.09, 8.11, 9.02, 9.06, 9.09)
pitch_array = {white_keys, black_keys, Cmaj7, Gmaj7, Amin7, Dmin7}
```

```
pitch_length = len(pitch_array)
pitch = pitch_array[trand(0, pitch_length)]

wave1 = maketable("wave3", 10, 0,1, 1,1, 2,1, 3,0, 4,0.5, 5,1)
wave2 = maketable("wave3", 1000, 0,1, 1,1, 2,1, 3,0, 4,0.5, 5,1)
wave3 = maketable("wave", 1000, "sine")
wave4 = maketable("wave", 1000, "saw99")
wave5 = maketable("wave", 1000, "tri")
wave6 = maketable("random", 20, "even", -1,1)
wave7 = maketable("wave", 10, "buzz")
wave8 = maketable("wave", 10, "saw")
wave9 = maketable("wave", 10, "sine")
waveform_array = {wave1, wave2, wave3, wave4, wave5, wave6, wave7, wave8, wave9}
waveform_length = len(waveform_array)
waveform = waveform_array[trand(0, waveform_length)]

panL = maketable("random", 10, "low", 0,0.6)
panR = maketable("random", 10, "high", 0.4,1)

increment = 0.25
for(start = 0; start < 25; start += increment){
     rate = rate_array[trand(0, rate_length)]
     }

//left channel wavetable
for(start = 0; start < 525; start += increment){
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     envelope = envelope_array[trand(0, envelope_length)]
     pitch = pitch_array[trand(0, pitch_length)]
     waveform = waveform_array[trand(0, waveform_length)]
     panL = maketable("random", 10, "low", 0,0.6)
     WAVETABLE(start, duration, amplitude*envelope, cpspch(pitch), panL)
     }

//right channel wavetable
for(start = 0; start < 525; start += increment){
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     envelope = envelope_array[trand(0, envelope_length)]
     pitch = pitch_array[trand(0, pitch_length)]
```

```
     waveform = waveform_array[trand(0, waveform_length)]
     panR = maketable("random", 10, "low", 0,0.6)
     WAVETABLE(start, duration, amplitude*envelope, cpspch(pitch), panR)
     }

//---------------------------MODALBAR
start = 0
duration = 0.02  //seconds
amplitude = 30000  //absolute btwn 0-32768
pitch = 7.00
hardness = 1.0
position = 1.0
instrument = 4 //Agogo is 2
pan = 0.5

stagger = 0.01
increment = 0.25

for(start = 0; start < 525; start += increment){
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     pitch = pitch_array[trand(0, pitch_length)]
     hardness = irand(0.2, 1.0)
     MMODALBAR(start,  duration, amplitude, cpspch(pitch), hardness, position,
     instrument = 4, 0)
     MMODALBAR(start+stagger, duration, amplitude, cpspch(pitch), hardness, position,
     instrument, 1)
     stagger += 0.5
     pitch = pitch - 1
     pan = trunc(random())
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     MMODALBAR(start+stagger, duration, amplitude, cpspch(pitch), hardness, position,
     instrument, pan)
     instrument = pickrand(1, 2, 3, 4, 5, 6, 7, 8)
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     MMODALBAR(start, duration, amplitude, cpspch(pitch), hardness, position, instrument,
     pan)
     duration += 0.0002
     increment = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
```

```
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     MMODALBAR(start, duration, amplitude, cpspch(pitch), hardness, position, instrument,
     pan)
     }


//Day 35
rtsetparams(44100, 2)
load("WAVETABLE")
load("FREEVERB")
load("NOISE")
load("MOOGVCF")
load("MMODALBAR")
srand()
bus_config("WAVETABLE", "aux 0-1 out")
bus_config("FREEVERB", "aux 0-1 in", "out 0-1")
bus_config("NOISE", "aux 2-3 out")
bus_config("MOOGVCF", "aux 2-3 in", "out 0-1")

fibonacci_array = {1, 1, 2, 3, 5, 8, 13, 21, 34, 55, 89, 144, 233}
fibonacci_length = len(fibonacci_array)
total_duration = 360

//-----------------------------WAVETABLE
start = 0
duration = 5 //slight overlap
amplitude = 100
envelope = maketable("line", 1000, 0,0, 0.5,1, 1.0,0)
pan = 0.5
pitch_array = {"C4", "D4", "E4", "F4", "G4", "A4", "B4", "C5"}
pitch_length = len(pitch_array)

for(start = 0; start < total_duration; start += 1){
     pitch = pitch_array[trand(0, pitch_length)]
     pan = random()
     WAVETABLE(start,   duration, amplitude*envelope, cpslet(pitch), pan)
     WAVETABLE(start+1, duration, amplitude*envelope, (cpslet(pitch) - irand(0.5,5)),
pan)
     }

//-------------------Long reverb
start = 0
instart = 0
```

```
duration = total_duration
amplitude = 0.9
room = 0.9
predelay = 0.05
ringdur = 10
damp = 30
dry = 40
wet = 90
width = 100
FREEVERB(start, instart, duration, amplitude, room, predelay, ringdur, damp, dry, wet,
width)

//------------------------------NOISE
start = 0
duration = total_duration
amplitude = 10000
envelope = maketable("line", 1000, 0,0, 0.6,0.7, 1,0)
pan = makeLFO("sine", 10.5, 0,1)
NOISE(start, duration, amplitude*envelope, pan)

//------------------------------MOOGVCF (for NOISE)
start = 0
instart = 0
duration = total_duration
amplitude = 0.3
envelope = maketable("random", 25, "gaussian", 0,1)
inchan = 0
pan = makeLFO("sine", 0.25, 0,1)
bypass = 0
lowcf = 50
highcf = 1000
lowres = 0.7
highres = 1.0
cf = maketable("line", "nonorm", 2000, 0,lowcf, 10*.2,lowcf, 10*.5,highcf, 10,lowcf)
res = maketable("line", "nonorm", 2000, 0,lowres, 1,highres, 2,lowres)
MOOGVCF(start, instart, duration, amplitude*envelope, inchan, pan, bypass, cf, res)

//-------------------Long reverb
start = 0
instart = 0
duration = total_duration
amplitude = 0.9
```

```
room = 0.9
predelay = 0.05
ringdur = 10
damp = 30
dry = 40
wet = 90
width = 100
FREEVERB(start, instart, duration, amplitude, room, predelay, ringdur, damp, dry, wet,
width)

//---------------------------MODALBAR
start = 0
duration = 0.02  //seconds
amplitude = 30000  //absolute btwn 0-32768
pitch = 7.00
hardness = 1.0
position = 1.0
instrument = 4 //Agogo is 2
pan = 0.5

stagger = 0.01
increment = 0.75

for(start = 0; start < total_duration; start += increment){
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 40)
     pitch = pitch_array[trand(0, pitch_length)]
     hardness = irand(0.2, 1.0)
     MMODALBAR(start, duration, amplitude, cpslet(pitch), hardness, position,
     instrument = 4, 0)
     MMODALBAR(start+stagger, duration, amplitude, cpslet(pitch), hardness, position,
     instrument, 1)
     stagger += 0.5
     pan = trunc(random())
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     MMODALBAR(start+stagger, duration, amplitude, cpslet(pitch), hardness, position,
     instrument, pan)
     instrument = pickrand(1, 2, 3, 4, 5, 6, 7, 8)
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     MMODALBAR(start, duration, amplitude, cpslet(pitch), hardness, position, instrument,
```

```
     pan)
     duration += 0.0002
     increment = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     duration = (fibonacci_array[trand(0, fibonacci_length)] / 10)
     amplitude = (fibonacci_array[trand(0, fibonacci_length)] * 10)
     MMODALBAR(start,  duration, amplitude, cpslet(pitch), hardness, position,
     instrument, pan)
     }

//Day 36
rtsetparams(44100, 2)    //EXTRA thanks to Joel Matthys for looping assistance
load("STRUM2") //karplus strong algorithm
load("REVERBIT")
load("STRUM")  //older kp instrument
load("FLANGE")
load("MOOGVCF")
srand()

bus_config("STRUM2", "aux 0-1 out")
bus_config("REVERBIT", "aux 0-1 in", "out 0-1")
bus_config("STRUM", "aux 2-3 out")
bus_config("FLANGE", "aux 2-3 in", "aux 4-5 out")
bus_config("MOOGVCF", "aux 4-5 in", "out 0-1")

Cmaj7 = pickrand(7.00, 7.04, 7.07, 7.11, 8.00, 8.04, 8.07, 8.11)
Gmaj7 = pickrand(7.07, 7.11, 8.02, 8.06, 8.07, 8.11, 9.02, 9.06)
Amin7 = pickrand(7.09, 8.00, 8.04, 8.07, 8.09, 9.00, 9.04, 9.07)
Bmin7 = pickrand(7.11, 8.02, 8.06, 8.09, 8.11, 9.02, 9.06, 9.09)
pitch_array = {Cmaj7, Gmaj7, Amin7, Bmin7}
pitch_length = len(pitch_array)
pitch = pitch_array[trand(0, pitch_length)]
pitch_length = len(pitch_array)

//------------------------STRUM2
start = 0
duration = 0.125
amplitude = 20000
pitch = pitch_array[1]
squish = 1.0    //how squishy is the plucking medium (think hard pick to finger pad)
decay = 1.0    //seconds
pan = random()
```

```
for(iteration = 0; iteration < 100; iteration += 1){
     for(index = 0; index < pitch_length; index += 1){
               pitch = pitch_array[index]
               pan = pickrand(0, 1)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + 0.125  //change durations here
          }

     for(index = index-1; index >= 0; index -= 1){
               pitch = pitch_array[index]
               pan = pickrand(0, 1)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + 0.125
          }
     }

//--------------REV
start = 0
instart = 0
duration = 60
amplitude = 0.8
envelope = maketable("line", 1000, 0,0, 0.612,1.0, 1.0,0)
revtime = maketable("line", 1000, 0,0.01, 0.5,0.1, 1.0,0.2) //keep these short
revamnt = 0 //0 is dry, 1 is wet
chandelay = 0.1 //R channel delay to L channel
cutoff = 10000 //LOP~ cutoff in Hz
REVERBIT(start, instart, duration, amplitude*envelope, revtime, revamnt, chandelay, c
utoff)

//-------------STRUM (older, requires makegen and comes in several flavors
start = 0
duration = 0.25
pitch = pitch_array[1]
fundamentaldecaytime = 1.0
nyquistdelaytime = 0.1
amplitude = 5000
squish = 1.0

for(iteration = 0; iteration < 50; iteration += 1){

     for(index = 0; index < pitch_length; index += 1){
               pitch = pitch_array[index]
```

```
                pan = pickrand(0, 1)
                START(start, duration, pitch, fundamentaldecaytime, nyquistdelaytime,
                amplitude, squish, pan)
                start = start + 0.25  //change durations here
           }

     for(index = index-1; index >= 0; index -= 1){
                pitch = pitch_array[index]
                pan = pickrand(0, 1)
                START(start, duration, pitch, fundamentaldecaytime, nyquistdelaytime,
                amplitude, squish, pan)
                start = start + 0.25
           }
     }

//--------------------------FLANGE
start = 0
instart = 0
flangedur = 60
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, 0.305,1.0, 1.0,0)
resonance = maketable("line", 1000, 0,0.2, 0.5,0.9, 1.0,0.2)
maxdelay = 5.0
lowpitch = cpspch(9.09)
moddepth = maketable("line", 1000, 0,10, 1,50)
modrate = maketable("line", 1000, 0,20, 1,12000)
wetdrymix = maketable("line", 1000, 0,0, 1,1)
panfrequency = 0.13// makeconnection("inlet", 3, 0.5)
pan = makeLFO("sine", panfrequency, 0,1)
ringdur = 0
waveform = maketable("wave3", 1000, 1,1,45)
FLANGE(start, instart, flangedur, amplitude*envelope, resonance, maxdelay, moddepth,
modrate, wetdrymix,"FIR", 0, pan, ringdur, waveform)

//-----------------------------MOOGVCF
start = 0
instart = 0
duration = 60
amplitude = 0.3
inchan = 0
pan = makeLFO("sine", 0.25, 0,1)
bypass = 0
```

```
lowcf = 50
highcf = 1000
lowres = 0.7
highres = 1.0
cf = maketable("line", "nonorm", 2000, 0,lowcf, 10*.2,lowcf, 10*.5,highcf, 10,lowcf)
res = maketable("line", "nonorm", 2000, 0,lowres, 1,highres, 2,lowres)
MOOGVCF(start, instart, duration, amplitude, inchan, pan, bypass, cf, res)

//Day 37
rtsetparams(44100, 2)
load("STRUM2") //karplus strong algorithm
load("REVERBIT")
load("STRUMFB")     //karplus with feedback and distortion
load("FLANGE")
load("MOOGVCF")
srand()

bus_config("STRUM2", "aux 0-1 out")
bus_config("REVERBIT", "aux 0-1 in", "out 0-1")
bus_config("STRUMFB", "aux 2-3 out")
bus_config("FLANGE", "aux 2-3 in", "aux 4-5 out")
bus_config("MOOGVCF", "aux 4-5 in", "out 0-1")

Cmaj7 = pickrand(7.00, 7.04, 7.07, 7.11, 8.00, 8.04, 8.07, 8.11)
Gmaj7 = pickrand(7.07, 7.11, 8.02, 8.06, 8.07, 8.11, 9.02, 9.06)
Amin7 = pickrand(7.09, 8.00, 8.04, 8.07, 8.09, 9.00, 9.04, 9.07)
Dmin7 = pickrand(7.02, 7.05, 7.09, 8.00, 8.02, 8.05, 8.09, 9.00)
pitch_array = {Cmaj7, Gmaj7, Amin7, Dmin7}
pitch_length = len(pitch_array)
pitch = pitch_array[trand(0, pitch_length)]
pitch_length = len(pitch_array)
//------------------------STRUM2
start = 0
duration = 0.25
amplitude = 20000
pitch = pitch_array[1]
squish = 1.0    //how squishy is the plucking medium (think hard pick to finger pad)
decay = 1.0    //seconds
pan = random()

for(iteration = 0; iteration < 100; iteration += 1){
```

```
     for(index = 0; index < pitch_length; index += 1){
               pitch = pitch_array[index]
               pan = pickrand(0, 1)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + 0.125  //change durations here
          }

     for(index = index-1; index >= 0; index -= 1){
               pitch = pitch_array[index]
               pan = pickrand(0, 1)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + 0.125
          }
     }
//--------------REV
start = 0
instart = 0
duration = 60
amplitude = 0.8
envelope = maketable("line", 1000, 0,0, 0.612,1.0, 1.0,0)
revtime = maketable("line", 1000, 0,0.01, 0.5,0.1, 1.0,0.2) //keep these short
revamnt = 0 //0 is dry, 1 is wet
chandelay = 0.1 //R channel delay to L channel
cutoff = 10000 //LOP~ cutoff in Hz
REVERBIT(start, instart, duration, amplitude*envelope, revtime, revamnt, chandelay,
cutoff)
//-------------STRUMFB same as STRUM, but with feedback and decay
start = 0
duration = 0.25
amplitude = 5000
pitch = pitch_array[1]
feedbackpitch = pitch - 0.5
squish = 1.0
fundamentaldecaytime = 1.0
nyquistdecaytime = 0.1
distortiongain = 5.0
feedbackgain = 0.05
cleanlevel = 0
distortionlevel = 1.0

for(iteration = 0; iteration < 100; iteration += 1){
```

```
     for(index = 0; index < pitch_length; index += 1){
               pitch = pitch_array[index]
               pan = pickrand(0, 1)
               STRUMFB(start, duration, amplitude, pitch, feedbackpitch, squish,
               fundamentaldecaytime,
               nyquistdecaytime,distortiongain, feedbackgain, cleanlevel,
               distortionlevel)
               start = start + 0.25  //change durations here
          }

     for(index = index-1; index >= 0; index -= 1){
               pitch = pitch_array[index]
               pan = pickrand(0, 1)
               STRUMFB(start, duration, amplitude, pitch, feedbackpitch, squish,
               fundamentaldecaytime, nyquistdecaytime,distortiongain, feedbackgain,
               cleanlevel, distortionlevel)
               start = start + 0.25
          }
     }
//--------------------------FLANGE
start = 0
instart = 0
flangedur = 60
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, 0.305,1.0, 1.0,0)
resonance = maketable("line", 1000, 0,0.2, 0.5,0.9, 1.0,0.2)
maxdelay = 5.0
lowpitch = cpspch(9.09)
moddepth = maketable("line", 1000, 0,10, 1,50)
modrate = maketable("line", 1000, 0,20, 1,12000)
wetdrymix = maketable("line", 1000, 0,0, 1,1)
pan = makeLFO("sine", 13.1, 0,1)
ringdur = 0
waveform = maketable("wave3", 1000, 1,1,45)
FLANGE(start, instart, flangedur, amplitude*envelope, resonance, maxdelay, moddepth,
modrate, wetdrymix,"FIR", 0, pan, ringdur, waveform)
//-----------------------------MOOGVCF
start = 0
instart = 0
duration = 60
amplitude = 0.3
inchan = 0
```

```
pan = makeLFO("sine", 0.25, 0,1)
bypass = 0
lowcf = 500
highcf = 2000
lowres = 0.2
highres = 1.0
cf = maketable("line", "nonorm", 2000, 0,lowcf, 10*.2,lowcf, 10*.5,highcf, 10,lowcf)
res = maketable("line", "nonorm", 2000, 0,lowres, 1,highres, 2,lowres)
MOOGVCF(start, instart, duration, amplitude, inchan, pan, bypass, cf, res)

//Day 38
rtsetparams(44100, 2)
/*
Start by seeding the random number generator to CPU clock time
*/
srand()
/*
There are a few ways to generate random numbers in RTcmix
1. rand() will generate a random number between -1 and 1, useful for random waveforms
2. random() will generate a random number between 0 and 1, good for mapping later
3. irand() and trand() are one and the same, or so it seems, and allow a range to be cho-
sen
*/
random_number_a = rand()
random_number_b = random()
random_number_x = irand(0, 10) // or trand() I suppose...
printf("1: %f \n", random_number_a)
printf("2: %f \n", random_number_b)
printf("3. %f \n", random_number_x)
/*
trand() or irand() returns a large number like 2.60620117188, so we could truncate it or
round it...
*/
printf("The raw value of our random number is: %f \n", random_number_x)
x_truncated = trunc(random_number_x)
x_rounded = round(random_number_x)
printf("Random number: %f \n", random_number_x)
printf("Truncated, it equals: %f \n", x_truncated)
printf("Rounded, it equals: %f \n", x_rounded)
/*
The only weighted distribution I know for now is taking the average of two random
numbers, or
a "triangle" distribution.
```

```
*/
random_number_x = round(trand(0, 10))
random_number_y = round(trand(0, 10))
average_of_random_numbers = ((random_number_x + random_number_y) / 2)
print(random_number_x)
print(random_number_y)
printf("The average is: %f \n", average_of_random_numbers)
/*
So now, I could do this in a loop to get a wide variety of numbers, which will most
likely point to numbers near the middle of 0 and 10. I think.
*/
for(i = 0; i < 10; i += 1){
     print_off()
     random_number_x = round(trand(0, 10))
     random_number_y = round(trand(0, 10))
     average_of_random_numbers = ((random_number_x + random_number_y) / 2)
     print_on()
     print(average_of_random_numbers)
     }
/*
RTcmix has a built in makerandom() that will do the same thing, or a maketable() of
random numbers
*/
triangular_distribution1 = makerandom("triangle", 10, 0,10) //10 values/sec
triangular_distribution2 = maketable("random", 10, "triangle", 0,10)
//plottable(triangular_distribution2)
/*
So Joel Matthys has a library of functions that will do some of the things found most
commonly in the C library of math functions. sin, cos, etc., but I'm interested in map,
which works like the [expr_scale] object in Pure Data.
*/
load("jfuncs")
print(sin(1))
print(cos(1))
print(lowrand())    //all fit in the range of 0-1
print(highrand())
print(trirand())
print(gaussrand())
print(constrain(5.3235,4.2,-98.543))
print(map(2,1,4,50,100))
print(prob(0.5,0.1))
/*
```

```
Using these, it's possible to generate random numbers from any given range
*/
srand()  //these aren't reacting to srand()?
value = highrand() // random number between 0 and 1, weighted high
print(value)
value_mapped = map(value, 0,1, 0,25)  //like [expr_scale], fit in range 0-25
print(value_mapped)
/*
Filling some panning values with a table of random numbers between 0 and 1, weighted
toward R
*/
pan = maketable("random", 100, "high", 0,1)
//plottable(pan)
/*
Now put it to use in the PAN() instrument for more precise control
*/
load("WAVESHAPE")
load("PAN")
print_off()

bus_config("WAVESHAPE", "aux 0-1 out")
bus_config("PAN", "aux 0-1 in", "out 0-1")

//--------------------------------WAVESHAPE
start = 0
duration = 20
frequency = makeLFO("sine", 0.1, 40,22100)
mindistortionindex = 0.0
maxdistortionindex = 1.0
amplitude = 20000
envelope = makerandom("prob", 10, -1.0,1.0,0,0)
pan = 0.5  //pan later
wave = maketable("wave", 1000, "sine")
transferfunc = maketable("cheby", 1000, 0.9, 0.3,-0.2,0.6,-0.7)
indexguide = maketable("line", 1000, 0,0, 3.5,1, 7,0)
WAVESHAPE(start, duration, frequency, mindistortionindex, maxdistortionindex,
amplitude*envelope, pan, wave, transferfunc, indexguide)

//--------------------------------PAN
start = 0
instart = 0
duration = 20
```

```
amplitude = maketable("line", 1000, 0,10.0, 0.8,0.5, 1.0,0)
inputchannel = 0
pantype = 1  //0 for constant power, 1 for linear
pan = maketable("random", 10, "high", 0,1)
//plottable(pan)
PAN(start, instart, duration, amplitude, inputchannel, pantype, pan)
/*
Now, modtable(normalize) will do the same thing as map, but with elements in a table
*/
random_numbers = maketable("random", "nonorm", 10, "gaussian", 0,50)
//important to include "nonorm"
plottable(random_numbers)
table_mod = modtable(random_numbers, "normalize", 4000)  //constrain between 0.0 and 4000
plottable(table_mod)

//---------------------------------WAVETABLE
load("WAVETABLE")
start = 20
duration = 10
amplitude = 3000
frequency = table_mod
pan = modtable(random_numbers, "normalize", 1)
WAVETABLE(start, duration, amplitude, frequency, pan)


//Day 39
rtsetparams(44100, 2)    //A drum beat
load("WAVETABLE")
load("PAN")
load("REV")
load("FLANGE")
load("JCHOR")
control_rate(4)
print_off()
srand()

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("WAVETABLE", "out 0-1")

bus_config("PAN", "aux 0-1 in", "aux 2-3 out")
bus_config("PAN", "aux 0-1 in", "aux 4-5 out")
bus_config("REV", "aux 2-3 in", "out 0-1")
```

```
bus_config("FLANGE", "aux 4-5 in", "aux 6-7 out")
bus_config("JCHOR", "aux 6-7 in", "out 0-1")

totalduration = 360
//----------------------Bass Drum 1
start = 0
bd_duration = 1
bd_amplitude = 20000
bd_envelope = maketable("line", 1000, 0,1.0, 0.8,1.0, 1.0,0)
bd_frequency = 70
bd_pan = 0.5
bd_wave = maketable("wave", 1000, "sine")

bd_loop = 2
for(start = 0; start < totalduration; start += bd_loop){
     WAVETABLE(start, bd_duration, bd_amplitude, bd_frequency, bd_pan, bd_wave) //BD1

     if(start >= 36){
          WAVETABLE(start, bd_duration, bd_amplitude-15000, bd_frequency*2, bd_pan,
          bd_wave)
          }

     if(start >= 48){
          WAVETABLE(start, bd_duration, bd_amplitude-19999, bd_frequency*24, bd_pan,
          bd_wave)
          }

     }

//----------------------Hi Hat 1
start = 8
hh_duration = 0.01
hh_amplitude = 60000
hh_frequency = 3000
hh_pan = random()
hh_wave = maketable("random", 5, "high", -1,1)

hh_loop = 0.5
for(start = 8; start < totalduration; start += hh_loop){
     WAVETABLE(start, hh_duration, hh_frequency, hh_pan = random(), hh_wave) //HH1
     }
```

```
//-----------------------Snare
start = 16
sd_duration = 0.025
sd_amplitude = 10000
sd_frequency = 54052
sd_pan = makeLFO("sine", 8.0, 0,1)
sd_wave = maketable("wave", 1000, "buzz")

sd_loop = 1
for(start = 16; start < totalduration; start += sd_loop){

     if(start%2 == 0){
          WAVETABLE(start+sd_loop, sd_duration, sd_amplitude, sd_frequency, sd_pan,
          sd_wave) //SD
          }

     }

//-----------------------Hi Hat 2
start = 24
hh_duration = 0.001
hh_amplitude = 15000
hh_frequency = 30000
hh_pan = random()
hh_wave = maketable("random", 5, "high", -1,1)

hh_loop = 0.25
for(start = 24; start < totalduration; start += hh_loop){
     WAVETABLE(start, hh_duration, hh_frequency, hh_pan = random(), hh_wave) //HH2
     }

//-----------------------Bass Drum 2
start = 36
bd_duration = 0.75
bd_amplitude = 0.2
bd_frequency = 630
bd_pan = random()
bd_wave = maketable("wave", 1000, "sine", 1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1,1)
//lots of partials
bd_loop = 2
for(start = 36; start < totalduration; start += bd_loop){
```

```
     WAVETABLE(start, bd_duration, bd_frequency*3, bd_pan = random(), bd_wave) //BD2
     }

//----------------------PAN
start = 36
instart = 0
duration = totalduration
amplitude = 10.0
envelope = maketable("line", 1000, 0,0, 0.4,1.0, 0.9,1.0, 1.0,0)
inchannel = 0
panmode = 1        //0 for constant power pan, 1 for linear pan
pan = makeLFO("sine", 4.25, 0,1)
PAN(start, instart, duration, amplitude*envelope, inchannel, panmode, pan)

//-----------------------REV
start = 36
instart = 0
duration = totalduration
amplitude = 10.0
type = 1     // 1 is Perry Cook's, 2 is John Chowning's, 3 is Michael McNabb's
rvbtime = 0.25
rvbpct = 0.05
inchan = 0
REV(start, instart, duration, amplitude, type, rvbtime, rvbpct, inchan)

//--------------------FLANGE
start = 44
instart = 0  //again, 0 from auxiliary bus
duration = totalduration
flamplitude = 40.0
flenvelope = maketable("random", 100, "low", 0.3,1.0)
resonance = irand(0.3,1.0)
maxdelay = 5.0
moddepth = maketable("line", "nonorm", 100, 0,1, 5,5, 10,7)
modspeed = maketable("line", 1000, 0,0.1, 1,0.9)
wetdrymix = 0.8
filtertype = "IIR"
inputchannel = 0
pan = makeLFO("sine", 4.0, 0,1)
ringdownduration = lowrand()
flangewave = maketable("wave", 2000, "sine")
FLANGE(start, instart, duration, flamplitude*flenvelope, resonance, maxdelay,
```

```
moddepth, modspeed, wetdrymix, filtertype, inputchannel, pan, ringdownduration,
flangewave)

//--------------------CHORUS
start = 44
instart = 0.00
outdur = totalduration
indur = 0.01
maintain_dur = 1
transposition = -0.05
nvoices = 10
minamp = 0.1
maxamp = 0.5
minwait = 0.00
maxwait = 0.60
seed = srand()
setline(0,0, .5,1, outdur/8,1, outdur,0)
makegen(2, 7, 1000, 0, 10, 1, 990, 0)
JCHOR(start, instart, outdur, indur, maintain_dur, transposition, nvoices, minamp,
maxamp, minwait, maxwait, seed)
```

```
# Day 40
# RTcmix not only uses MINC as its parser, but also Python, a powerful programming
# language. There are some differences in style (take for example the commenting...), but
# both MINC and Python work well in creating new score files.

# Remember, when executing RTcmix scripts in the Terminal, you need to call the command
# cmix followed by the < symbol. (cmix < /path/to/scorefile.sco) In order to use Python,
# you first need to configure RTcmix to interpret Python scrips (done by running
# ./configure --with-python during installation), and then call
# pycmix < /path/to/scorefile.sco in the Terminal window.

# First, pycmix score files need the following command to run...

from rtcmix import *

# which allows Python to use the list of cmix commands. (I *think* Python comes
# preloaded on Mac OSX)

#Then the header of your file looks the same as it has using MINC
```

```
rtsetparams(44100, 2)
load("WAVETABLE")

#-----------------------------WAVETABLE
start = 0
duration = 5
amplitude = 90  #in dB
pitch = 7.09    #octave point pitch class
pan = 0.5
wavetype = maketable("wave", 1000, "sine")
WAVETABLE(start, duration, ampdb(amplitude), pitch, pan, wavetype)

# The above preceding code produces a five second sine wave on the note A.
# Now, it's possible to set a loop (looks a little different than MINC) and
# generate some random pitches.

start = 5
duration = 1
amplitude = 90  #in dB
pitch = 7.09    #octave point pitch class
pan = 0.5
wavetype = maketable("wave", 1000, "sine")

for start in range(5, 23):
     print_off()
     pitch = pickrand(7.00, 7.01, 7.02, 7.03, 7.04, 7.05, 7.06, 7.07, 7.08, 7.09, 7.10,
7.11)
     WAVETABLE(start, duration, ampdb(amplitude), pitch, pan, wavetype)
     print_on()
     print(midipch(pitch))

# This type of loop uses range( ), which is intrisnic to Python. In this case with
# range(5, 23) we are asking for a loop that STARTS at 5, counts up incrementally to 22
# by 1. There are other ways to use range, such as range(0, 101, 2) which counts from 0-
# 100 by 2 instead of by 1.# Conditional statements in Python also look a bit different.



# Day 41
from rtcmix import *

rtsetparams(44100, 2)
load("WAVETABLE")
```

```
#-----------------------------WAVETABLE
start = 0
duration = 0.25
amplitude = 90  #in dB
pitch = 7.09    #octave point pitch class
pan = 0.5
wavetype = maketable("random", 16, "gaussian", -1,1)

# Getting through loops with a floating point number. (Is there an easier way?)
# Similar to how I've been using increment or loop as variables in MINC loops...
# i.e - loop = 0.25; for(start = 0; start < 10; start += loop)

def drange(start, stop, step):
     r = start
     while r < stop:
          yield r
          r+= step

for i in drange(0.0, 11.0, 0.25):
     pitch = pickrand(7.00, 7.01, 7.02, 7.03, 7.04, 7.05, 7.06, 7.07, 7.08, 7.09, 7.10,
     7.11)
     WAVETABLE(i, duration, ampdb(amplitude), pitch, pan, wavetype)

     random_number = trunc(irand(0, 10))

     if random_number == 5:
          pan = 0

     elif random_number < 5:
          pan = 0

     else:
          pan = 1

# Day 42
# Checking out arrays in Python, help from John Gibson's 02.13.2001 help file

from rtcmix import *

rtsetparams(44100, 2)
load("WAVETABLE")
```

```
#-----------------------------WAVETABLE
start = 0
duration = 0.0625
amplitude = 90  #in dB
pitch_array = [5.00, 5.001, 5.02, 5.03, 5.05, 5.07, 5.069, 5.10, 6.00]
pitch_array_length = len(pitch_array)
pan = 0.5
wavetype = maketable("random", 16, "gaussian", -1,1)

def drange(start, stop, step):
     r = start
     while r < stop:
          yield r
          r+= step

for i in drange(0.0, 11.0, duration):
     slot = int(random() * pitch_array_length * .999999)
     current_pitch = pchoct(pitch_array[slot])

     WAVETABLE(i, duration, ampdb(amplitude), current_pitch, pan, wavetype)
     random_number = trunc(irand(0, 10))

     if random_number == 5:
          pan = 0

     elif random_number < 5:
          pan = 0.5
          pitch = ampdb(9.01)

     else:
          pan = 1


//Day 43
/*
The [rtcmix~] object in Pure Data now includes the ability to play sound from data stored
in an array. It requires two messages...

1. First, a "bufset nameofarray" message needs to be banged FIRST. This points the
[rtcmix~] object to the array in question. So, if my array is called 'array1' - or the
name of the default array name in Pd - my message will thus be "bufset array1". If my
```

```
array is called 'foo', I'll say "bufset foo", etc.

2. Lastly, your rtinput( ) message needs to say "PDBUF" and "nameofarray". Example below
*/

rtinput("PDBUF", “array1")  //necessary "PDBUF" and "nameofarray", in this case 'array1'.

//--------------------------STEREO for playback (MIX is another useful instrument here)
bus_config("STEREO", "in 0-1", "out 0-1")

start = 0
instart = 0
//your duration matches the length of 'sound' in your array. Here, I'm writing to
// the array for ten seconds.
duration = 10
amplitude = 1.0
pan = 0.5
STEREO(start, instart, duration, amplitude, pan)

/*
...now, to play it backwards...
*/

//----------------------------SCRUB
bus_config("SCRUB", "in 0-1", "out 0-1")
start = 10
instart = 10
//to play backwards, set instart to end of file, in this case ten seconds
duration = 10
amplitude = 1.0
speed = -1.0
//play backwards, 1.0 normal, 0.5 1/2 speed, -0.5 backwards 1/2 speed, 2.0 2X as fast...
sync_width = 16
oversampling = 40        //16 and 40 seem to be good defaults
inchannel = 0
pan = 0.5
SCRUB(start, instart, duration, amplitude, speed, sync_width, oversampling, inchannel,
pan)

//--------------------------STEREO again
start = 20
instart = 0
```

```
duration = 0.75
amplitude = 1.0
pan = random()

loop = 0.5          //slight overlap for each iteration

for(start = 20; start < 100; start += loop){
     instart = irand(0.25,5.0)
     pan = random()
     STEREO(start, instart, duration, amplitude, pan)
     }

// Day 44
from rtcmix import *
rtsetparams(44100, 2)
load("STEREO")
load("TRANS")
control_rate(44100)

srand()
rtinput(“/path/to/file.wav”)

#---------- STEREO
start = 0
instart = 0
duration = 0.25
envelope = maketable("line", 1000, 0,0, 0.1,1.0, 0.9,1.0, 1.0,0)

#---------- TRANS
amplitude = maketable("spline", 1000, 50, 0,0.8, 10,0.9, 50,0.5, 100,0.9)

def drange(start, stop, step):
     x = start
     while x < stop:
          yield x
          x+= step

increment = 1.0

for start in drange(0,1000,increment):
     instart = irand(0,DUR())
```

```
    pan = random()
    duration = pickrand(0.25, 1.25, 0.75, 0.125, 0.33, 2.5)
    STEREO(start, instart, duration, 1.0 * envelope , pan)

    instart = irand(0,DUR())
    pan = random()
    duration = pickrand(0.25, 1.25, 0.75, 0.125, 0.33, 2.5)
    STEREO(start, instart, duration, 1.0 * envelope, pan)

    low = octpch(-0.11)
    high = octpch(0.11)
    transpcurve = maketable("line", 100, 0,0, 0.5,high, 1.0,low)
    transpositions = makeconverter(transpcurve, "pchoct")
    duration = pickrand(0.25, 1.25, 0.75, 0.125, 0.33)
    TRANS(start+instart, start, duration, amplitude * envelope, transpositions)

    increment = pickwrand(0.25,10, 0.0125,20, 1.0,30, 2.0,20, 0.125,20)


//Day 45
rtsetparams(44100, 2)
load("HALFWAVE")
load("REVERBIT")
srand(8)
totalduration = 360

//------------------------HALFWAVE
start = 0
duration = 1.75

octavegamut = {5, 6, 7}
octavegamut_length = len(octavegamut)

pitchgamut = {0.00, 0.01, 0.04, 0.05, 0.06, 0.08, 0.10, 0.11}
pitchgamut_length = len(pitchgamut)

amplitude = 5000
envelope = maketable("line", 100, 0,0, 2,1, 9,0)
wave1 = maketable("wave3", 1000, 3.14,1,0, 6.28,1,0.5)
wave2 = maketable("wave3", 1000, 1.00,1,0, 2.00,1,0.5)
wave3 = maketable("wave3", 1000, 1,1,0, 3,0.3,0, 5,0.2,0, 7,0.05,0, 9,0.01,0, 11,0.001,0)
wave4 = maketable("wave3", 1000, 1,1,0, 2,0.5,0, 3,0.3,0, 4,0.25,0, 5,0.2,0, 6,0.16,0,
```

```
7,0.14,0, 8,0.125,0)
wave5 = maketable("wave3", 1000, 1,1,0, 3,0.14,0, 5,0.04,0, 7,0.02,0, 9,0.012,0,
11,0.008,0)
wavegamut = {wave1, wave2, wave3, wave4, wave5}
wavegamutlength = len(wavegamut)

wavecrossoverpoint = 0.5
pan = 0.5

increment = 0.5

for(start = 0; start < totalduration; start += increment){
     octave = octavegamut[trand(0,octavegamut_length)]
     nextpitch = pitchgamut[trand(0,pitchgamut_length)]
     pitch = nextpitch + octave

     if(octave == 2){
          duration = 5.0
          pan = makeLFO("sine", 1.25, 0,1)
          HALFWAVE(start, duration, cpspch(pitch)+0.04, amplitude*envelope, wave1, wave2,
               wavecrossoverpoint, pan)
          }

     wave1 = wavegamut[trand(0,wavegamutlength)]
     wave2 = wavegamut[trand(0,wavegamutlength)]

     random_number_x = round(trand(0, 10))
     random_number_y = round(trand(0, 10))
     average_of_random_numbers = ((random_number_x + random_number_y) / 2)
     wavecrossoverpoint = average_of_random_numbers / 10

     pan = random()
     HALFWAVE(start, duration, cpspch(pitch), amplitude*envelope, wave1, wave2,
     wavecrossoverpoint, pan)
     }


//Day 46
//building waveform from an array

amplitudes_array = {1, 2, 3, 4, 5, 6, 7, 8, 9, 10}
```

```
for(i = 0; i < 10; i += 1){
     randomnumber_x = irand(-1,1)
     randomnumber_y = irand(-1,1)
     partial_amplitude = (randomnumber_x + randomnumber_y) / 2

     amplitudes_array[i] = partial_amplitude
     }

waveform = maketable("wave3", 31,    1, amplitudes_array[1], 0,
                                  2, amplitudes_array[2], 0,
               3, amplitudes_array[3], 0,
               5, amplitudes_array[4], 0,
               8, amplitudes_array[5], 0,
               13,amplitudes_array[6], 0,
               21,amplitudes_array[7], 0,
               34,amplitudes_array[8], 0,
               55,amplitudes_array[9], 0,
               89,amplitudes_array[10],0)

plottable(waveform)
```

```
//Day 47
rtsetparams(44100, 2)
load("HALFWAVE")
srand()

amplitudes_array = {1, 2, 3, 4, 5, 6, 7, 8, 9, 10}

for(i = 0; i < 10; i += 1){
     randomnumber_x = irand(-1,1)
     randomnumber_y = irand(-1,1)
     partial_amplitude = (randomnumber_x + randomnumber_y) / 2

     amplitudes_array[i] = partial_amplitude
     }

//------------------------HALFWAVE
start = 0
duration = 10

octavegamut = {2, 10, 11}
octavegamut_length = len(octavegamut)
```

```
pitchgamut = {0.00, 0.01, 0.04, 0.05, 0.06, 0.08, 0.10, 0.11}
pitchgamut_length = len(pitchgamut)

octave = octavegamut[trand(0,octavegamut_length)]
nextpitch = pitchgamut[trand(0,pitchgamut_length)]
pitch = nextpitch + octave

amplitude = 1000
waveform1 = maketable("wave3", 31,  1, amplitudes_array[1], 0,
               3, amplitudes_array[2], 0,
               5, amplitudes_array[3], 0,
               5, amplitudes_array[4], 0,
               7, amplitudes_array[5], 0,
               8, amplitudes_array[6], 0,
               11,amplitudes_array[7], 0,
               14,amplitudes_array[8], 0,
               15,amplitudes_array[9], 0,
               20,amplitudes_array[10],0)
waveform2 = makefilter(waveform1, "invert", 0)
wavecrossoverpoint = 0.5
pan = 0.5

HALFWAVE(start, duration, cpspch(pitch), amplitude, waveform1, waveform2,
wavecrossoverpoint, pan)

//Day 48
rtsetparams(44100, 2)
load("HALFWAVE")
reset(8)
srand()

amplitudes_array = {1, 2, 3, 4, 5, 6, 7, 8, 9, 10}

for(i = 0; i < 10; i += 1){
     randomnumber_x = irand(-1,1)
     randomnumber_y = irand(-1,1)
     partial_amplitude = (randomnumber_x + randomnumber_y) / 2

     amplitudes_array[i] = partial_amplitude
     }
```

```
//------------------------HALFWAVE
start = 0
duration = 360

octavegamut = {1, 2, 3}
octavegamut_length = len(octavegamut)

pitchgamut = {0.00, 0.01, 0.04, 0.05, 0.06, 0.08, 0.10, 0.11}
pitchgamut_length = len(pitchgamut)

octave = octavegamut[trand(0,octavegamut_length)]
nextpitch = pitchgamut[trand(0,pitchgamut_length)]
pitch = nextpitch + octave

amplitude = 1000
waveform1 = maketable("wave3", 100, 1, amplitudes_array[1], -0.5,
               2, amplitudes_array[2], 1.0,
               6, amplitudes_array[3], 0.25,
               7, amplitudes_array[4], 0.0,
               9, amplitudes_array[5], -0.125,
               10,amplitudes_array[6], -0.33,
               11,amplitudes_array[7], 1.0,
               14,amplitudes_array[8], -1.0,
               15,amplitudes_array[9], 0.0,
               40,amplitudes_array[10],0.25)
wavecrossoverpoint1 = maketable("line", 10, 0,0, 1.0,1.0)
pan1 = makeLFO("sine", 0.05, 0.0,0.6)

waveform2 = makefilter(waveform1, "invert", 0)
wavecrossoverpoint2 = maketable("line", 10, 0,1.0, 1.0,0)
pan2 = makeLFO("sine", 0.25, 0.4,1.0)

HALFWAVE(start, duration, cpspch(2.11), amplitude, waveform1, waveform2,
wavecrossoverpoint1, pan1)
HALFWAVE(start, duration, cpspch(2.10)+0.01, amplitude, waveform1, waveform2,
wavecrossoverpoint2, pan2)



//Day 49
rtsetparams(44100, 2)
load("WAVETABLE")
load("DELAY")
```

```
load("JFUNCS")
rate = pickrand(2, 4, 8, 16, 32, 128, 512, 1024, 22051, 44051)
control_rate(8)
seed = srand()

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("DELAY", "aux 0-1 in", "out 0-1")

//-------------------------------WAVETABLE
start = 0
total_duration = 360

maxdb = 90
mindb = 62
amplitude = ampdb(maxdb-mindb)
envelope = maketable("curve", 1000, 0,1,-5, 1,0)

maxfrequency = 44052
minfrequency = 22051
waveform = maketable("wave", 1000, "sine")

maxincrement = 0.4
minincrement = 0.05
increment = 0.1

for (start = 0; start < total_duration; start += increment){
     duration = increment * 0.2
     amplitude = ampdb(irand(mindb, maxdb))
     frequency = irand(minfrequency, maxfrequency)
     glissfactor = irand(0.2, 0.7)

     if (random() < 0.5){     // 50% probability of glissing up
          glissfactor += 1
          }

     gliss = maketable("curve", "nonorm", 1000, 0,1,-2, 1,glissfactor)
     pan = random()
     WAVETABLE(start, duration, amplitude*envelope, frequency*gliss, pan, waveform)
     WAVETABLE(start, duration, amplitude*envelope, frequency*gliss*1.01, 1-pan,
     waveform)

     increment = irand(minincrement, maxincrement)
```

```
     control_rate(rate)
     }

//---------------------------------DELAY
start = 0
instart = 0
duration = total_duration
amplitude = 1.0
envelope = maketable("window", 1000, "hanning")
deltime = makerandom("even", 0.5, 0.1, 2.2, seed)
delaytime = makefilter(deltime, "smooth", lag = 0)
feedback = maketable("line", 1000, 0,0.18, 0.5,0.62, 1.0,0.02)
ringdur = 0.5     // seconds to ring out delay line after note is finished
inputchannel = 0
pan = 1
DELAY(start, instart, duration, amplitude*envelope, delaytime, feedback, ringdur,
inputchannel, pan)

deltime = makerandom("low", 2.5, 0.1, 3.3, seed)
delaytime = makefilter(deltime, "smooth", lag = 0)
inputchannel = 1
pan = 0
DELAY(start, instart, duration, amplitude*envelope, delaytime, feedback, ringdur,
inputchannel, pan)

//Day 50
rtsetparams(44100, 2)
load("MIX")
load("REV")
print_off()

rtinput(“/path/to/file.wav”)
bus_config("MIX", "in 0-1", "aux 0-1 out")
bus_config("REV", "aux 0-1 in", "out 0-1")

totalduration = 176 * 2

//-----------------------------MIX
start = 0
instart = 0
duration = 0.5
amplitude = 0.5
```

```
left = 0
right = 1

increment = 1.0

for(start = 0; start < totalduration; start += increment){
     instart = irand(1,176)
     left = irand(0,0.5)
     right = irand(0.5,1.0)
     MIX(start, instart, duration, amplitude, left,right)
     duration += 0.02
     }

//-----------------------------REV
start = 0
instart = 0
duration = totalduration
amplitude = 1.0
type = 3
rvbtime = 0.15
rvbpct = 0.08
inchan = 0

REV(start, instart, duration, amplitude, type, rvbtime, rvbpct, inchan)

//Day 51
rtsetparams(44100, 2)
load("STRUM2")

Cmajor = {7.00, 7.02, 7.04, 7.05, 7.07, 7.09, 7.11, 8.00}
arraylength = len(Cmajor)

//------------------------STRUM2
start = 0
duration = 0.5
amplitude = 20000
pitch = Cmajor[0]
squish = 1.0    //how squishy is the plucking medium (think hard pick to finger pad)
decay = 1.0    //seconds
pan = random()

transposition = 0.01
```

```
for(iteration = 0; iteration < 12; iteration += 1){

     for(index = 0; index < arraylength; index += 1){
               pitch = Cmajor[index] + transposition
               pan = pickrand(0, 1)
                STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + 0.33  //change durations here
          }

     for(index = index-1; index >= 0; index -= 1){
               pitch = Cmajor[index] + transposition
               pan = pickrand(0, 1)
               STRUM2(start, duration, amplitude, pitch, squish, decay, pan)
               start = start + 0.33
          }

      transposition += 0.01
     }

//Day 52
rtsetparams(44100, 2)
load("SCRUB")
load("STEREO")

rtinput(“/path/to/file.wav”)

//----------------------SCRUB meta
start = 0
instart = 0
amplitude = 1.0
width = 16
oversampling = 40
inchannel = 0

//-----tape head 1
durationL = 1.0
speedL = 1.0
panL = 1.0          //RTcmix panning is L = 1.0, center = 0.5, R = 0.0

//-----tape head 2
speedR = 0.90
```

```
panR = 0.0
durationR = durationL * speedR

for(start = 0; start < 100; start += 1.5){
     SCRUB(start, instart, durationL, amplitude, speedL, width, oversampling, inchannel,
     panL)
     SCRUB(start, instart, durationR, amplitude, speedR, width, oversampling, inchannel,
     panR)
     }

//Day 53
rtsetparams(96000, 2)
load("TRANS")
load("REVERBIT")

bus_config("TRANS", "in 0-1", "aux 0-1 out")
bus_config("REVERBIT", "aux 0-1 in", "out 0-1")

rtinput(“/path/to/file.wav”)

//--------------------TRANS
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
transposition = maketable("line", 1000, 0,0.0, 0.5,-0.2, 1.0,-1.0)
inchannel = 0
pan = 0.5
TRANS(start, instart, duration, amplitude, transposition, inchannel, pan)

//--------------------REVERBIT(so it sounds like its in my glass mantle)
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
reverbtime = 5.8
reverbpct = 0.9
channeldelay = 0.1
cutoff = 500
REVERBIT(start, instart, duration, amplitude, reverbtime, reverbpct, channeldelay, cut-
off)
```

```
//Day 54
rtsetparams(44100, 2)
load("WAVETABLE")
load("MROOM")
control_rate(44100)

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("MROOM", "aux 0-1 in", "out 0-1")

soprano = {7.07, 7.09, 7.11, 8.00, 8.02, 8.00, 7.11, 8.07, 8.06, 8.04, 8.02, 8.00, 7.09,
7.06, 7.07}
alto = {7.02, 7.02, 7.02, 7.04, 7.02, 7.02, 7.02, 7.07, 7.09, 7.07, 7.07, 7.07, 7.04,
7.02, 7.02}
tenor = {6.11, 6.09, 6.07, 6.07, 6.09, 6.06, 6.07, 6.07, 7.02, 6.07, 6.07, 6.07, 6.07,
7.00, 6.11}
bass = {5.07, 5.06, 5.07, 5.04, 5.06, 6.02, 5.07, 5.11, 5.09, 6.00, 5.11, 6.04, 6.00,
6.02, 5.07}

//------------------------WAVETABLE
start = 0
duration = 2
amplitude = 10000
envelope = maketable("window", 1000, "hanning")
pitch = 8.00
pan = random()
waveform = maketable("wave", 1000, "sine")

placeholder = 1

for(start = 0; start < 15 * duration; start += duration){
     pan = random()

WAVETABLE(start, duration, amplitude*envelope, soprano[placeholder]+1,random(), wave-
form)
WAVETABLE(start, duration, amplitude*envelope, alto[placeholder]+1, random(), wave-
form)
WAVETABLE(start, duration, amplitude*envelope, tenor[placeholder]+1, random(), wave-
form)
WAVETABLE(start, duration, amplitude*envelope, bass[placeholder]+1, random(), wave-
form)

     placeholder += 1
     }

//-------------------------CHURCH HALL
start = 0
instart = 0
duration = 30
```

```
amplitude = 0.5
xdim = 500      //distance (feet) from middle of room to right wall
ydim = 800      //distance (feet) from middle of room to front
rvbtime = 25.0
reflect = 90.0  //amount reflected by walls
innerwidth = 4.0
inchan = 0
quantizationrate = 96000

timeset(0,          0-xdim, 0-ydim)
timeset(duration,   xdim,   ydim)

setline(0,0, duration/8,1, duration-.5,1, duration,0)
MROOM(start, instart, duration, amplitude, xdim, ydim, rvbtime, reflect, innerwidth,
inchan, quantizationrate)

//Day 55
rtsetparams(44100, 2)
load("WAVETABLE")
load("FREEVERB")
srand()

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("FREEVERB", "aux 0-1 in", "out 0-1")

soprano = {7.07, 7.09, 7.11, 8.00, 8.02, 8.00, 7.11, 8.07, 8.06, 8.04, 8.02, 8.00, 7.09,
7.06, 7.07}
alto = {7.02, 7.02, 7.02, 7.04, 7.02, 7.02, 7.02, 7.07, 7.09, 7.07, 7.07, 7.07, 7.04,
7.02, 7.02}
tenor = {6.11, 6.09, 6.07, 6.07, 6.09, 6.06, 6.07, 6.07, 7.02, 6.07, 6.07, 6.07, 6.07,
7.00, 6.11}
bass = {5.07, 5.06, 5.07, 5.04, 5.06, 6.02, 5.07, 5.11, 5.09, 6.00, 5.11, 6.04, 6.00,
6.02, 5.07}

//------------------------WAVETABLE
start = 0
totalduration = 1000
amplitude = 10000
envelope = maketable("window", 1000, "hanning")
pitch = 8.00
pan = random()
waveform = maketable("wave", 1000, "sine")
```

```
sopranoplaceholder = 0
altoplaceholder = 0
tenorplaceholder = 0
bassplaceholder = 0

sopranoduration = 10          //tendency to speed up
altoduration = 10        //starts fast and relaxes
tenorduration = 11       //pretty even keel, cool person
bassduration = 14        //strangely slow heart rate overall

for(start = 0; start < totalduration; start += sopranoduration){
     sopranoduration += irand(0,0.6)
     WAVETABLE(start, sopranoduration, amplitude*envelope, soprano[sopranoplaceholder]+1,
     random(),waveform)
     sopranoplaceholder += 1
     }

for(start = 0; start < totalduration; start += altoduration){
     altoduration -= irand(0,0.01)
     WAVETABLE(start, altoduration, amplitude*envelope, alto[altoplaceholder]+1,
     random(), waveform) altoplaceholder += 1
     }

for(start = 0; start < totalduration; start += tenorduration){
     tenorduration -= irand(0,0.01)
     WAVETABLE(start, tenorduration, amplitude*envelope, tenor[tenorplaceholder]+1,
     random(), waveform) tenorplaceholder += 1
     }

for(start = 0; start < totalduration; start += bassduration){
     bassduration += irand(-0.3,0.5)
     WAVETABLE(start, bassduration, amplitude*envelope, bass[bassplaceholder]+1,
     random(), waveform) bassplaceholder += 1
     }

//-----------------------------FREEVERB
start = 0
instart = 0
duration = totalduration
amplitude = 0.3
room = 1.0
```

```
predelay = 0.0001
ringdur = 15
damp = 80
dry = 10
wet = 90
width = 100
FREEVERB(start, instart, duration, amplitude, room, predelay, ringdur, damp, dry, wet,
width)

//Day 56
rtsetparams(44100, 2)
load("NOISE")
load("MULTICOMB")
load("PAN")
control_rate(44100)
srand()

bus_config("NOISE", "aux 0-1 out")
bus_config("MULTICOMB", "aux 0-1 in", "aux 2-3 out")
bus_config("PAN", "aux 2-3 in", "out 0-1")

//----------------------NOISE
start = 0
duration = 0.009
amplitude = 50000
envelope1 = maketable("line", 1000, 0,0, 5,1, 10,0)
envelope2 = maketable("line", 1000, 0,0, 7,0.5, 9,1, 10,0)
envelope3 = maketable("line", 1000, 0,0, 0.1,1, 0.5,0.05, 10,0)
pan = random()

//----------------------MULTICOMB
start = 0
instart = 0
combdur = duration
combamp = 0.5
combenv = maketable("line", 1000, 0,0, 0.1,0.8, 0.8,0.5, 1.0,0)
low = cpspch(13.01)
high =cpspch(13.05)
revtime = 0.1 //seconds

//-----------------------PAN
start = 0
```

```
instart = 0
duration = duration
panamp = 1.0
inchannel = 0
panmode = 1        //0 for constant power pan, 1 for linear pan
pan = random()

loop = 0.001

for (start = 0; start < 250; start += loop){
     NOISE(start0.01, duration, amplitude*envelope3, random())
     MULTICOMB(start, instart, duration+0.5, combamp*combenv, low, high, revtime)
     PAN(start, instart, duration, panamp, inchannel, panmode, random())

     step = irand(-0.2,0.3)
     loop += step

     if(loop >= 0.1){
          duration += 0.01
          revtime += 0.01
          }
     }

//Day 57
rtsetparams(44100, 2)
load("WIGGLE")
srand()

//-----------------------WIGGLE
start = 0
duration = 8.2
amplitude = maketable("line", "nonorm", 1000, 0,0, 0.1,2000, 5,4000, 10,2000)
envelope = maketable("curve", 2000, 0,0,2, 2.5,1,0, 13,1,-3, 25,0)

min = 2.00
max = 12.00
gliss = maketable("random", "nonorm", "nointerp", 500, "low", min, max)
pitch = 18.09
freq = makeconverter(octpch(pitch) + gliss, "cpsoct")

mod_depth_type = 1 // 0 is no mod, 1 is % of carrier, 2 is FM
filt_type = 2      // 0 is no filt, 1 is low, 2 is high
```

```
filt_steep = 2
balance = true     // balance output and input signals
carrierwaveform = maketable("wave", 24051, "sine")
mod_wavetable = maketable("wave", 1000, "sine")
mod_freq = 200
mod_depth = 20
filt_cf = maketable("curve", "nonorm", 2000, 0,1000,-4, 1,1)
pan = random()

for(start = 0; start < 100; start += 5){
     pan = random()
     freq -= 10.2
     WIGGLE(start, duration, amplitude*envelope, freq, mod_depth_type, filt_type,
     filt_steep, balance, carrierwaveform, mod_wavetable, mod_freq, mod_depth, filt_cf,
     pan)
     }

//Day 58
rtsetparams(44100, 2)
load("GRANSYNTH")
seed = srand()
//---------------------GRANSYNTH
start = 0
duration = 10
amplitude = 10000
waveform = maketable("wave", 1000, "tri")
genv = maketable("window", 1000, "hanning")
ghop = maketable("random", 1000, "low", 0.01, 1.0)
goutjitter = 0.01
gdurmin = 0.0001  // longer durations = "pitchy"
gdurmax = 0.009  // shorter are "clicky"
gampmin = 0.2
gampmax = 1.0
gpitches = {15.01, 19.03, 19.09, 19.11, 21.0, 23.08}
gplength = len(gpitches)
gtrans = maketable("literal", "nonorm", 0, 0.01, 0.03, 0.11, 1.0) // octpcs
gtransjitter = 1.0 // maximum random amount to shift current pitch
panmin = 0.0
panmax = 1.0

loop = 1.0 // for loop
```

```
for (start = 0; start < 25; start += loop){
     pitchindex = trand(gplength)
     gpitch = gpitches[pitchindex]
     GRANSYNTH(start, duration, amplitude, waveform, genv, ghop, goutjitter, gdurmin,
     gdurmax, gampmin, gampmax, gpitch, gtrans, gtransjitter, seed, panmin, panmax)
     goutjitter -= 0.01
     gpitch += 30
     }

//Day 59
rtsetparams(44100, 2)
load("WAVETABLE")
load("REVERBIT")

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("REVERBIT", "aux 0-1 in", "out 0-1")

//--------------------dates
age = 30
month = 7.00
date = 19

//--------------------meta
amplitude = age * 1000
envelope = maketable("line", 1000, 1,1, 9,1, 10,0)
pan = makeLFO("sine", age/100, 0,1)
waveform = maketable("wave3", 1000, 1,1,0, age/10,1,0, 5,1,0, month,1,0, date,1,0,
age,1,0)

//--------------------melody in the key of "F"
pitchgamut = {month, month+.02, month+.04, month+.05, month+.07, month+.09, month+.10,
month+1.00}

//--------------------durations
teenth = 0.275
sixdot = 0.6
quatre = 1.0

WAVETABLE(0, sixdot, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(0.625,      teenth, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(1,     quatre, amplitude*envelope, cpspch(pitchgamut[1]), pan, waveform)
WAVETABLE(2,     quatre, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
```

```
WAVETABLE(3,     quatre, amplitude*envelope, cpspch(pitchgamut[3]), pan, waveform)
WAVETABLE(4,     quatre, amplitude*envelope, cpspch(pitchgamut[2]), pan, waveform)
WAVETABLE(6,     sixdot, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(6.625, teenth, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(7,     quatre, amplitude*envelope, cpspch(pitchgamut[1]), pan, waveform)
WAVETABLE(8,     quatre, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(9,     quatre, amplitude*envelope, cpspch(pitchgamut[4]), pan, waveform)
WAVETABLE(10,    quatre, amplitude*envelope, cpspch(pitchgamut[3]), pan, waveform)
WAVETABLE(12,    sixdot, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(12.625,teenth, amplitude*envelope, cpspch(pitchgamut[0]), pan, waveform)
WAVETABLE(13,    quatre, amplitude*envelope, cpspch(pitchgamut[7]), pan, waveform)
WAVETABLE(14,    quatre, amplitude*envelope, cpspch(pitchgamut[5]), pan, waveform)
WAVETABLE(15,    sixdot, amplitude*envelope, cpspch(pitchgamut[3]), pan, waveform)
WAVETABLE(15.625,teenth, amplitude*envelope, cpspch(pitchgamut[3]), pan, waveform)
WAVETABLE(16,    quatre, amplitude*envelope, cpspch(pitchgamut[2]), pan, waveform)
WAVETABLE(17,    quatre, amplitude*envelope, cpspch(pitchgamut[1]), pan, waveform)
WAVETABLE(19,    sixdot, amplitude*envelope, cpspch(pitchgamut[6]), pan, waveform)
WAVETABLE(19.625,teenth, amplitude*envelope, cpspch(pitchgamut[6]), pan, waveform)
WAVETABLE(20,    quatre, amplitude*envelope, cpspch(pitchgamut[5]), pan, waveform)
WAVETABLE(21,    quatre, amplitude*envelope, cpspch(pitchgamut[3]), pan, waveform)
WAVETABLE(22,    quatre, amplitude*envelope, cpspch(pitchgamut[4]), pan, waveform)
WAVETABLE(23,    quatre+1, amplitude*envelope, cpspch(pitchgamut[3]), pan, waveform)

//-------------------REVERBIT
start = 0
instart = 0
duration = age
amplitude = age/100
reverbtime = month+date
reverbpct = age/100+age/100+age/100
channeldelay = age/100
cutoff = age*100
REVERBIT(start, instart, duration, amplitude, reverbtime, reverbpct, channeldelay,
cutoff)

//Day 60
rtsetparams(44100, 2)
load("MIX")
load("COMBIT")
srand()

rtinput(“/path/to/file.wav”)
```

```
bus_config("MIX", "in 0-1", "aux 0-1 out")
bus_config("COMBIT", "aux 0-1 in", "out 0-1")

totalduration = DUR()

//-----------------------------MIX
start = 0
instart = 0
duration = 0.5
amplitude = 0.1
envelope = maketable("line", 1000, 0,0, 0.5,1, 1.0,0)
left = 0
right = 1

increment = 1.0

for(start = 0; start < totalduration; start += increment){
     instart = irand(0,30)
     MIX(start, instart, duration, amplitude*envelope, left,right)
     }

//----------------------------COMBIT
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
ringtime = 1.1
inchan = 0

COMBIT(start, instart, duration, amplitude, cpspch(7.11), ringtime, inchan, random())
COMBIT(start, instart, duration, amplitude, cpspch(8.07), ringtime, inchan, random())
COMBIT(start, instart, duration, amplitude, cpspch(8.09), ringtime, inchan,
makeLFO("sine", 4.075, 0,1))

//Day 61
rtsetparams(44100, 2)
load("VOCODESYNTH")
srand()

rtinput(“/path/to/file.aiff”)
```

```
//---------------------------VOCODESNTH
start = 0
instart = 0
duration = DUR()
amplitude = 15.0
envelope = maketable("line", 1000, 0,0, .1,0.5, 0.6,0.5, 1.0,0)
numbands = 50
lowcf = maketable("random", 1000, "gaussian", 300,500)
interval = 0.07
cartransp = 0.03
bw = maketable("random", 100, "low", 0,1) / 100

winlen = 0.2
smooth = 0.98
thresh = 0.0001
atktime = 0.001
reltime = 0.01
hipassmod = 0.03
hipasscf = maketable("line", 1000, 0,1000, 1.0,3000)
spacemult = cpspch(interval) / cpspch(0.0)
inchannel = 0
pan = 0.5
carwavetable = maketable("wave", 1000, "sine")
scale1 = 0.5
scale2 = 1.0
scalecurve = maketable("curve", "nonorm", 1000, 0,scale1,1, 1,scale2)

VOCODESYNTH(start, instart, duration, amplitude*envelope, numbands, lowcf, spacemult,
cartransp, bw,winlen, smooth, thresh, atktime, reltime, hipassmod, hipasscf, inchannel,
pan, carwavetable, scalecurve)

//Day 62
rtsetparams(44100, 2)
load("FMINST")
srand()

start = 0
duration = 0.05
amplitude = 5000
carrier = cpspch(7.00)
modulatorfrequency = 220
minindex = 10
```

```
maxindex = 30
pan = random()
waveform = maketable("wave", 1000, "sine")
guide = maketable("line", "nonorm", 1000, 0, 0, 5,1, 7, 0)

increment = 0.125

for(start = 0; start < 100; start += increment){
     duration = irand(0.01, 0.1)

     FMINST(start, duration, amplitude, carrier, modulatorfrequency, minindex, maxindex,
     pan = random(), waveform, guide)
     modulatorfrequency = carrier * 22051
     carrier = start * 0.14
     }



//Day 63
rtsetparams(44100, 2)
load("JGRAN")
srand()
//----------------------JGRAN
start = 0
duration = 2.0
amplitude = 1.0
randomseed = srand()
oscconfig = 0 //  FM
oscphase = 0  // randomize osc phase? 0 is no; 1 is yes
grainenv = maketable("window", 1000, "hanning")
grainwaveclickhi = maketable("wave", 1000, "sine")
grainwaveclicklow = maketable("wave", 1000, "tri")
FMmult = maketable("random", 1000, "gaussian", 2.0, 100.0)
FMindex = maketable("line", 1000, 0,1.0, 1.0,30.0)
minfreq = 30
maxfreq = 22051
minspeed = maketable("line", "nonorm", 1000, 0,0.2, 1,0.001) // decreasing minimum
maxspeed = maketable("line", "nonorm", 1000, 0,0.001, 1,0.2) // increasing maximum
mindb = 0
maxdb = 90
density = maketable("random", 1000, "gaussian", 0,10)
pan = random()
```

```
panrand = maketable("random", 1000, "gaussian", 0,1)

loop = 0.0125

for (start = 0; start < 80; start += loop){
     JGRAN(start, duration, amplitude, randomseed, oscconfig, oscphase, grainenv,
     grainwaveclickhi,FMmult,FMindex, minfreq, maxfreq, minspeed, maxspeed, mindb, maxdb,
     density, pan = random(), panrand)
     }


//Day 64
rtsetparams(44100, 2)
load("STEREO")
load("PVOC")
load("PAN")

rtinput(“/path/to/file.aiff”)
bus_config("STEREO", "in 0-1", "aux 0-1 out")
bus_config("PVOC", "aux 0-1 in", "aux 2 out")
bus_config("PAN", "aux 2 in", "out 0-1")

totalduration = DUR()

//------------------------------STEREO
start = 0
instart = 0
duration = 1.0
amplitude = 0.5
envelope = maketable("line", 1000, 0,0.25, 0.2,1.0, 0.9,1.0, 1.0,0.25)
pan = 0.5

increment = 0.25

for(start = 0; start < totalduration; start += increment){
     instart = irand(0, totalduration)
     STEREO(start, instart, duration, amplitude*envelope, pan = random())
     duration += 0.02
     }

//------------------------------PVOC
start = 0
```

```
instart = 0
duration = totalduration
gain = 1.0
fftsize = 1024
winsize = fftsize * 2
decim = 1024
interp = 16
pitch = 0.0

PVOC(start, instart, duration, gain, 0, fftsize, winsize, decim, interp, pitch)

//-------------------------------PAN
start = 0
instart = 0
duration = 1.0
amplitude = 1.0
inchannel = 0
pantype = 1
pan = random()

for(start = 0; start < totalduration; start += increment){
     PAN(start, instart, duration, amplitude, inchannel, pantype, pan = random())
     }


//Day 65

srand()
print_off()
pitchgamut = {"C", "C#", "D", "Eb", "E", "F", "F#", "G", "Ab", "A", "Bb", "B", "C"}
octavegamut = {"3", "4", "5"}

octaveplaceholder = 0
pitchplaceholder = 0

for(start = 0; start < 15; start += 1){
     /*random_number_x = irand(0,11)
     random_number_y = irand(0,11)

     if(random_number_x >= random_number_y){
          pitchplaceholder = random_number_x + 1
          }
```

```
else{
     pitchplaceholder = random_number_y + 1
     }*/

//pitchplaceholder = ((random_number_x + random_number_y) / 2)

pitchplaceholder = irand(0,11)

random_number_a = irand(0,3) + 1
random_number_b = irand(0,3) + 1

octaveplaceholder = ((random_number_a + random_number_b) / 2)

//adjust pitches to stay within soprano sax range
/*if(octaveplaceholder < 1){
     pitchplaceholder += 1
     }*/

/*if(octaveplaceholder >= 2){
     pitchplaceholder += trunc(irand(0,3))
     }*/

     //now put them together
     octave = octavegamut[octaveplaceholder]
     pitch = pitchgamut[pitchplaceholder]

     nextpitch = pitch+octave
     print_on()
     print(nextpitch); print_off()
     }

//Day 66
rtsetparams(44100, 2)
load("AMINST")

start = 0
duration = 0.15
amplitude = 40000
carrierfrequency = 22051
modulatorfrequency = 44100
pan = random()
```

```
modulatoramplitude = maketable("line", 1000, 0,0, 1,1, 2,0)
carrierwave = maketable("wave", 1000, "sine")
modulatorwave = maketable("random", 20, "gaussian", -1,1)

increment = 0.45

for(start = 0; start < 250; start += increment){
     random_value = trand(0,10)
     if(random_value <= 6){
          carrierfrequency += 2
          modulatorfrequency += 4
          }
     if(random_value >= 6){
          carrierfrequency -= 3
          modulatorfrequency -= 2
          }
     pan = random()
     AMINST(start, duration, amplitude, carrierfrequency, modulatorfrequency,
               pan, modulatoramplitude, carrierwave, modulatorwave)
     increment = irand(0,1) / 2
     }
```

```
//Day 67
/*
spray_init( ) and get_spray( ) are similar to the Pd object [urn].
It returns a series of random numbers, but doesn't repeat a number that has been chosen.
Thus, "unrepeated random number."
*/

spray_table = 1
spray_size = 10
seed = srand()

spray_init(spray_table, spray_size, seed) // initialize table 3 with 7 elements

for (i = 0; i < spray_size; i += 1) {
     current_value = get_spray(spray_table) + 1
     print(current_value)
   }

//Day 68
rtsetparams(44100, 2)
```

```
load("MBRASS")

Cmajor = {7.00, 7.02, 7.04, 7.05, 7.07, 7.09, 7.11, 8.00}
arraylength = len(Cmajor)

//-------------------Trumpet-ish
start = 0
duration = 1
amplitude = 40000
pitch = Cmajor[0]
slide_length = 103
lip_filter = 140
max_pressure = 0.045
pan = 0.5
breath = maketable("line", 1000, 0,0, 0.05,1, 3.0,3, 3.5,0)

transposition = 0.01
loop = 0.125

for(iteration = 0; iteration < 12; iteration += 1){

for(index = 0; index < arraylength; index += 1){
          pitch = Cmajor[index] + transposition
          slide_length -= 4
          lip_filter += 10
          max_pressure += 0.155
          pan = pickrand(0, 1)
     MBRASS(start, duration, amplitude, pitch, slide_length, lip_filter,
     max_pressure, pan,breath)
          start += loop  //change durations here
     }

for(index = index-1; index >= 0; index -= 1){
          pitch = Cmajor[index] + transposition
          slide_length += 4
          lip_filter -= 10
          max_pressure -= 0.155
          pan = pickrand(0, 1)
     MBRASS(start, duration, amplitude, pitch, slide_length, lip_filter,
     max_pressure, pan, breath)
          start += loop
     }
```

```
     transposition += 0.01
     }

//Day 68 //Tweet script
rtsetparams(44100,2)load(“WAVETABLE")f=140g=maketable("expbrk",10,0,1,10,9)for(i=0;i<50;i
+=5){f+=i*20 WAVETABLE(i,20,1750,i*g,random())}

//Day 69 //Tweetable
rtsetparams(44100,2)
load("WAVETABLE")
load("JFUNCS")
reset(8)for(i=0;i<1000;i+=1){WAVETABLE(i,i/2,500,abs(sin(i)*220.5),sin(i))WAVETABLE(i/2,i
/3,500,abs(sin(i)*441),cos(i))}

//Day 70
rtsetparams(44100, 4)   //four channels
load("WAVETABLE")
load("NPAN")

bus_config("WAVETABLE", "aux 0 out")
bus_config("NPAN", "aux 0 in", "out 0-3")

//----------------------START
start = 0
duration = 30
amplitude = 10000
frequency = 440
pan = 0  //send everything through channel 0
waveform = maketable("random", 50, "gaussian", -1,1)
//noisy waveform to hear spatialization

for(start = 0; start < 25; start += 1){
     WAVETABLE(start, duration, amplitude, frequency, pan, waveform)
     frequency += irand(10,50)
     }

//---------------------NPAN
NPANspeakers("polar",
          45, 1,     // left front
          -45, 1,     // right front
          135, 1,     // left rear
          -135, 1)     // right rear
```

```
start = 0
instart = 0
duration = 60
amplitude = 1.0
mode = "xy" //or "polar"
x = maketable("random", 100, "high", -1,1)
y = maketable("random", 100, "low", -1,1)
NPAN(start, instart, duration, amplitude, mode, x, y)

//Day 71
rtsetparams(44100, 4)  //QPAN, similar to NPAN, but specific for quad sound
load("WAVETABLE")
load("QPAN")

bus_config("WAVETABLE", "aux 0 out")
bus_config("QPAN", "aux 0 in", "out 0-3")

//----------------------START
start = 0
duration = 30
amplitude = 10000
frequency = 440
pan = 0  //send everything through channel 0
waveform = maketable("random", 50, "gaussian", -1,1)
//noisy waveform to hear spatialization

for(start = 0; start < 25; start += 1){
     WAVETABLE(start, duration, amplitude, frequency, pan, waveform)
     frequency += irand(10,50)
     }

//----------------------QPAN
start = 0
instart = 0
duration = 60
amplitude = 1.0
x = maketable("random", 100, "high", -1,1)
y = maketable("random", 100, "low", -1,1)
QPAN(start, instart, duration, amplitude, x,y)
```

```
//Day 72
load("JFUNCS")

values_array = {380, 383, 660, 504, 512, 265, 212, 185, 184, 202, 204, 201, 978, 1008,
234}
num_values = len(values_array)

values_in_table = maketable("literal", "nonorm", (num_values * 2) + 1,
                            0, values_array[0],
                            1, values_array[1],
                            2, values_array[2],
                            3, values_array[3],
                            4, values_array[4],
                            5, values_array[5],
                            6, values_array[6],
                            7, values_array[7],
                            8, values_array[8],
                            9, values_array[9],
                            10,values_array[10],
                            11,values_array[11],
                            12,values_array[12],
                            13,values_array[13],
                            14,values_array[14])
plottable(values_in_table)

values_constrained = modtable(values_in_table, "normalize", 1.0)
plottable(values_constrained)

//"center" all values between the mean value
sum = 0

for(i = 0; i < num_values; i += 1){
     sum += values_array[i]
     }

mean_raw = sum / num_values
mean = map(mean_raw, 184,1008, 0,1)
print(mean)

//mean = 407.466, in this instance, scaled to 0.27 via map( )

//okay, so now have values "reflect" over the mean, creating a waveform
```

```
waveform = makefilter(values_constrained, "invert", mean)
//plottable(waveform)

//Day 73
include /path/to/a/previous/scorefile.sco

load("WAVETABLE")
srand()

//----------------------WAVETABLE
start = 0
duration = 3
amplitude = 10000
envelope = maketable("window", 1000, "hanning")
frequency = 0  //initialize
pan = random()
wave = waveform

for(start = 0; start < 30; start += 2){
     WAVETABLE(start, duration, amplitude*envelope, values_array[start / 2],
     pan = random(), wave)
     }

//Day 74
rtsetparams(44100, 2)
load("STEREO")
load("WAVESHAPE")
load("JFUNCS")
srand()

rtinput(“/path/to/file.aif”)

bus_config("STEREO", "in 0-1", "aux 0-1 out")
bus_config("WAVESHAPE", "aux 0-1 in", "out 0-1")

//take values and / 100 to get between 0 and 1
values_array = {.380, .383, .660, .504, .512, .265, .212, .185, .184, .202, .204, .201,
.978, 1.008, .234}
num_values = len(values_array)

//------------------------WAVESHAPE
start = 0
```

```
duration = 10
frequency = 0  //taken from array when in for() loop
min_distortion_index = 0.0
max_distortion_index = 1.0
amplitude = 3000.0
envelope = maketable("window", 1000, "hanning")
pan = random()

waveform = maketable("wave3", (num_values * 2) + 1,
                    0, values_array[0],0,
                    1, values_array[1],0,
                    2, values_array[2],0,
                    3, values_array[3],0,
                    4, values_array[4],0,
                    5, values_array[5],0,
                    6, values_array[6],0,
                    7, values_array[7],0,
                    8, values_array[8],0,
                    9, values_array[9],0,
                    10,values_array[10],0,
                    11,values_array[11],0,
                    12,values_array[12],0,
                    13,values_array[13],0,
                    14,values_array[14],0)

transferfunc = maketable("cheby", 1000, 0.9, 0.3,-0.2,0.6,-0.7)
indexguide = maketable("line", 1000, 0,0, 3.5,1, 7,0)

for(start = 0; start < 30; start += 2){
     WAVESHAPE(start, duration, (values_array[start / 2] * 2), min_distortion_index,
     max_distortion_index,amplitude*envelope, pan = random(), waveform, transferfunc,
     indexguide)
     }


//Day 75
rtsetparams(44100, 2)
load("WAVETABLE")
seed = srand()

//----------------------WAVETABLE
start = 0
```

```
duration = 10
amplitude = 500
envelope = maketable("line", 1000, 0,0, 0.1,1, 0.9,1, 1.0,0)
frequency = 7.00
pan = 0.5

loop = 1

for(start = 0; start < 20; start += loop){
     frequency = pickrand(8.00, 8.04, 8.07, 8.11)

     phase1 = makerandom("even", 1000, 0,1, seed)
     wavetype = maketable("wave3", 1000, 1,1,phase1)
     pan1 = random()
     WAVETABLE(start, duration, amplitude*envelope, frequency, pan1, wavetype)

     frequency = pickrand(8.00, 8.04, 8.07, 8.11)
     phase2 = (1.0 - phase1)
     pan2 = (1.0 - pan1)
     wavetype = maketable("wave3", 1000, 1,1,phase2)
     WAVETABLE(start, duration, amplitude*envelope, frequency, pan2, wavetype)
     if(pan1 <= 0.10){
          pan3 = makeLFO("sine", 1.25, 0,1)
          WAVETABLE(start, duration, amplitude*envelope, 6.00, pan3, wavetype)
          }
     }


//Day 76
rtsetparams(44100, 4)
load("MIX")
load("QPAN")
load("JFUNCS")
srand()

rtinput("/Users/JerodSommerfeldt/Desktop/rocks.wav")

bus_config("MIX", "in 0-1", "aux 0-1 out")
bus_config("QPAN", "aux 0-1 in", "out 0-3")

/*
```

```
Quad panning with tendency for front speakers to stay to the right
and rear speakers to stay to the left

[1]             [2]  speaker array

          me



[3]             [4]

Recall that RTcmix panning for QPAN looks like this

[-1]            [1]  front


        still me
          [0]

[-1]            [1]  rear
*/

//--------------- MIX
start = 0
instart = 0
duration = 0.125
amplitude = 1.0
pan = random()

//--------------- QPAN
start = 0
instart = 0 //reading from aux bus, must be 0
duration = 0.125
amplitude = 1.0
srcX = -1
srcY = 1

increment = 0.125

for(start = 0; start < 100; start += increment){
     instart = irand(0,DUR())
     MIX(start, instart, duration, amplitude, 0,1)
```

```
     randomtest1 = highrand()
     srcX = map(randomtest1, 0,1, -1,1)

     randomtest2 = lowrand()
     srcY = map(randomtest2, 0,1, -1,1)
     QPAN(start, instart = 0, duration, amplitude, srcX, srcY)
     }

//Day 77
rtsetparams(44100, 2)
load("MIX")
load("STEREO")

rtinput(“/path/to/file.wav”)

/*
Going for the "doubling" effect to boost signal, -0.0015 to 0.004 sec delay (Pellman)
*/

//------------- MIX
start = 0
instart = 0
duration = DUR()
amplitude = 0.9
MIX(start,         instart, duration, amplitude, 0,0)
MIX(start + 0.004, instart, duration, amplitude, 1,1)

/*
Compare with simply STEREO( )
*/
STEREO(start+DUR(), instart, duration, amplitude, 0.5)

//Day 78
rtsetparams(44100, 2)
load("DCBLOCK")
load("TRANS")
srand()

rtinput(“/path/to/file.wav”)

bus_config("DCBLOCK", "in 0-1", "aux 0-1 out")
bus_config("TRANS", "aux 0-1 in", "out 0-1")
```

```
//---------------- DCBLOCK
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
DCBLOCK(start, instart, duration, amplitude)

//---------------- TRANS
start = 0
instart = 0
duration = DUR()
amplitude = 0.8
//transposition
x = random()
y = random()
transpositionL = (((x + y) / 2) / 10)
transpositionR = ((1 - transpositionL) / 10)
panL = 0
panR = 1
TRANS(start, instart, duration, amplitude, transpositionL, inchan = 0, panL)
TRANS(start, instart, duration, amplitude, transpositionR, inchan = 1, panR)
```

```
//Day 79
/*
This is an example .c file for a very basic RTcmix function, which can be accessed and
used in any MINC score file. It will be called squareme and after creating and compiling
it in your RTcmix build, you'll be able to use it as myfunction(pfield1)

As advertised, all squareme will actually do is take an input number (x) and multiply it
by itself.
*/

//-------  1. Load header files to include in this function.
#include <stdlib.h>
#include <stdio.h>
#include <string.h>
#include <assert.h>
#include <ugens.h>
#include <math.h>

#define DEBUG //debug
```

```
//-------- 2. Define your function and declare what it will do.
double myfunction (float p[], int n_args, double pp[]){
   double inputvalue;
   inputvalue = (inputvalue * pp[0]);
   return(inputvalue);
}

//-------- 3. Create a profile for your function, so you can call on it in your score.
int profile(){
   UG_INTRO("squareme", squareme);
   return 0;
}

//Day 80
// Writing window functions for RTcmix, using C++. These would be used in
// maketable("window")

#include <stdlib.h>
#include <stdio.h>
#include <string.h>
#include <assert.h>
#include <ugens.h>

#define DEBUG

// John Gibson's Hanning and Hamming windows from table.cpp in RTcmix bin folder
// (lines 1384-1427)

static int _window_table(const Arg args[], const int nargs, double *array,
const int len){
     int window_type = 0;

     if (len < 2)
          return die("maketable (window)", "Table length must be at least 2.");

     if (nargs != 1)
          return die("maketable (window)", "Missing window type.");

     if (args[0].isType(StringType)) {
          if (args[0] == "hanning")
               window_type = 1;
```

```
     else if (args[0] == "hamming")
          window_type = 2;

     else if(args[0] == "rectangular")
          window_type = 3;

     else if(args[0] == "lanczos")
          window_type = 4;

     else
          return die("maketable (window)", "Unsupported window type \"%s\".",
          (const char *) args[0]);
}

else if (args[0].isType(DoubleType)) {
     window_type = (int) args[0];
}

else
     return die("maketable (window)",
                    "Window type must be string or numeric code.");

switch (window_type) {
case 1:         // hanning window
     for (int i = 0; i < len; i++){
          array[i] = -cos(2.0 * M_PI * (double) i / (double) len) * 0.5 + 0.5;
          }
     break;

case 2:         // hamming window
     for (int i = 0; i < len; i++) {
          double val = cos(2.0 * M_PI * (double) i / (double) len);
          array[i] = 0.54 - 0.46 * val;
          }
     break;

case 3: // rectangular window
     for (int i = 0; i < len; i++) {
          array[i] = 1;
          }
     break;
```

```
          case 4: // lanczos window
               for (int i = 0; i < len; i++){
                    array [i] = sinc(2.0 * (double) i / (double) len - 1.0);
                    }
               break;

          default:
               return die("maketable (window)", "Unsupported window type (%d).”,
               window_type);
          }

     return 0;
     }


//Day 81
srand()

//three coins, heads = 0 and tails = 1
coin1 = 0
coin2 = 1
coin3 = 0
a = 0
b = 0
c = 0

for(i = 0; i < 3; i += 1){
print_off()
coin1 = pickrand(0,1)
coin2 = pickrand(0,1)
coin3 = pickrand(0,1)

if(coin1 == 0){
     a = 2
     }
     else{
     a = 3
     }

if(coin2 == 0){
     b = 2
     }
```

```
     else{
     b = 3
     }

if(coin3 == 0){
     c = 2
     }
     else{
     c = 3
     }

sum = a + b + c
print_on()
printf("%d", sum);

if(sum%2 == 0){
     printf("--------");
     }
     else{
     printf("---  ---");
     }
}


//Day 82
// Generating "mixtures" in Stockausen's Studie II
rtsetparams(44100, 2)
load("WAVETABLE")

frequency = 690   //fundamental frequency in mixture 67, which starts the work
frequency_rounded = frequency

//These were all rounded due to Stockhausen's rounding of frequencies for his frequency tables.
row1_ratio = 1.07
row2_ratio = 1.14
row3_ratio = 1.21
row4_ratio = 1.29
row5_ratio = 1.38

frequency_array = {0, 1, 2, 3, 4}

for(i = 0; i < 5; i += 1){
```

```
     print(frequency_rounded)
     frequency *= row4_ratio  //this is where mixture 67 lies in tables of frequencies
     frequency_rounded = trunc(frequency + 0.5)
     frequency_array[i] = frequency_rounded
     }

//----------------  WAVETABLE, generating first mixture in the score, with envelope
start = 0
duration = 1.56167      //again, roughly. Stockhausen specifies 1 sec = 76.2 cm of tape
amplitude = ampdb(70)  //score is in dB and this gesture starts at -15
envelope = maketable("line", 100, 0,1.0, 1.0,0)  //looks to me like the slope is linear
frequency = 1  //only initializing here
pan = 0.5
waveform = maketable("wave", 1000, "sine")

WAVETABLE(start, duration, amplitude*envelope, frequency_array[0], pan, waveform)
WAVETABLE(start, duration, amplitude*envelope, frequency_array[1], pan, waveform)
WAVETABLE(start, duration, amplitude*envelope, frequency_array[2], pan, waveform)
WAVETABLE(start, duration, amplitude*envelope, frequency_array[3], pan, waveform)
WAVETABLE(start, duration, amplitude*envelope, frequency_array[4], pan, waveform)
print(amplitude)
```

```
//Day 83
// Adding reverb to previous score, for authenticity (albeit digital...)
rtsetparams(44100, 2)
load("WAVETABLE")
load("REV")

bus_config("WAVETABLE", "aux 0-1 out")
bus_config("REV", "aux 0-1 in", "out 0-1")

frequency = 690   //fundamental frequency in mixture 67, which starts the work
frequency_rounded = frequency

//These were all rounded due to Stockhausen's rounding of frequencies for his frequency
tables.
row1_ratio = 1.07
row2_ratio = 1.14
row3_ratio = 1.21
row4_ratio = 1.29
row5_ratio = 1.38

frequency_array = {0, 1, 2, 3, 4}
```

```
for(i = 0; i < 5; i += 1){
     print(frequency_rounded)
     frequency *= row4_ratio  //this is where mixture 67 lies in tables of frequencies
     frequency_rounded = trunc(frequency + 0.5)
     frequency_array[i] = frequency_rounded
     }

//----------------  WAVETABLE, generating first mixture in the score, with envelope
start = 0
mixture_duration= 1.56167
//again, roughly. Stockhausen specifies 1 sec = 76.2 cm of tape
amplitude = ampdb(70)  //score is in dB and this gesture starts at -15
envelope = maketable("line", 100, 0,0, 0.9,1.0, 1.0,0)  //trying it reversed
frequency = 1  //only initializing here
pan = 0.5
waveform = maketable("wave", 1000, "sine")

WAVETABLE(start, mixture_duration, amplitude*envelope, frequency_array[0], pan, waveform)
WAVETABLE(start, mixture_duration, amplitude*envelope, frequency_array[1], pan, waveform)
WAVETABLE(start, mixture_duration, amplitude*envelope, frequency_array[2], pan, waveform)
WAVETABLE(start, mixture_duration, amplitude*envelope, frequency_array[3], pan, waveform)
WAVETABLE(start, mixture_duration, amplitude*envelope, frequency_array[4], pan, waveform)

//------------------  REVERB, digital, so not super accurate
start = 0
instart = 0
rvbtime = 2.0
duration = mixture_duration * rvbtime
amplitude = 1.0
type = 1     // 1 is Perry Cook's, 2 is John Chowning's, 3 is Michael McNabb's
rvbpct = 0.5
inchan = 0
REV(start, instart, duration, amplitude, type, rvbtime, rvbpct, inchan)

//Day 84
/*
Pauline Oliveros' Sonic Rorschach truncated to nine minutes
*/
rtsetparams(44100, 2)
load("NOISE")    //white noise
load("WAVETABLE")  //loud, short pulse
```

```
//------------------------  NOISE
start = 0
noiseduration = 60 * 9
amplitude = 2000
envelope = maketable("line", 1000, 0,0, 0.01,1.0, 0.9,1.0, 1.0,0)
NOISE(start, noiseduration, amplitude*envelope)

//------------------------  WAVETABLE
start = noiseduration / 2
duration = 1.25
amplitude = 50000
frequency = 2000
pan = 0.5
waveform = maketable("wave", 1000, "buzz")
WAVETABLE(start, duration, amplitude, frequency, pan, waveform)

//Day 85
rtsetparams(44100, 2)
load("FILTSWEEP")
srand()
rtinput(“/path/to/file.aiff”)

start = 0
instart = 0
duration = DUR()
amplitude = 0.75
envelope = maketable("random", 10, "high", 0.1,1.0)

ringdur = 0.5
balance = 0
steepness = 1

lowcf = 30
highcf = 4000
narrowbw = -0.5
widebw = -0.0125

inputchannel = 0
pan = makeLFO("sine", 8.75, 0,1)
bypass = 0
```

```
cf = maketable("line", "nonorm", 1000, 0,lowcf, 0.5,highcf, 1.0,lowcf)
bw = maketable("random", 10, "even", narrowbw,widebw)
FILTSWEEP(start, instart, duration, amplitude*envelope, ringdur, steepness, balance,
inputchannel, pan, bypass, cf, bw)


//Day 86
rtsetparams(44100, 2)
load("IIR")
load("JFUNCS")

//-------------  PULSES
start = 0
duration = 0.075
amplitude = 40000
envelope = maketable("window", 1000, "hanning")
firstpitch = cpsmidi(100)
pan = random()

centerfreq = 400
bandwidth = 0.25
filteramp = 10.0

for(start = 0; start < 500; start = start + 0.1) {
     cf = highrand(start, centerfreq)
     bw = highrand(start / 10, bandwidth)
     setup(cf, bw, filteramp)
     pitch = sin(firstpitch)
     PULSE(start, duration, amplitude*envelope, abs(pitch), pan)
     }

//Day 87
rtsetparams(44100, 2)
load("MMODALBAR")

srand()

//-------------------Modal Bar parameters
start = 0
duration = 1.0
amplitude = 40000
note = 880
```

```
stickhardness = 0.5  //0.0-1.0
stickposition = 0.9  //0.0-1.0
instrument = 3

for(start = 0; start < 4 * 40; start += 4){
               pan = pickwrand(0,0.2, 1,0.8)

     MMODALBAR(start, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     MMODALBAR(start+.75, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     pan = pickwrand(0,0.2, 1,0.8)
     MMODALBAR(start+0.5, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     pan = pickwrand(0,0.2, 1,0.8)
     MMODALBAR(start+1, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     pan = pickwrand(0,0.2, 1,0.8)
     MMODALBAR(start+2, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     pan = pickwrand(0,0.2, 1,0.8)
     MMODALBAR(start+2.75, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     pan = pickwrand(0,0.2, 1,0.8)
     MMODALBAR(start+3.5, duration, amplitude, note, stickhardness, stickposition,
     instrument, pan)
     pan = pickwrand(0,0.2, 1,0.8)
     }

//Day 88
rtsetparams(44100, 2)
load("MSITAR")

//-------------  MSITAR
start = 0
duration = 1.25
amplitude = 20000
pluck_amp = 0.5
pan = 0.5
amplitude_envelope = maketable("line", 1000, 0,0, 0.5,1.0, 1.0,0)

//long hand MIDI-to-frequency conversion
```

```
midinote = 60
num_halfsteps = 24  //quarter-tones, now that 8ve is divided by 24
A4 = 440 //in Hz
frequency = (pow(2, ((midinote - 69) / num_halfsteps)) * A4)

loop = 0.125
for(iteration = 0; iteration < 50; iteration += 1){

     for(index = 0; index < num_halfsteps; index += 1){
               frequency = (pow(2, ((midinote - 69) / num_halfsteps)) * A4)
          MSITAR(start, duration, amplitude, frequency, pluck_amp, pan,
          amplitude_envelope)
          midinote += 1
          start += loop  //change durations here
          }

     for(index = index - 1; index >= 0; index -= 1){
               frequency = (pow(2, ((midinote - 69) / num_halfsteps)) * A4)
          MSITAR(start, duration, amplitude, frequency, pluck_amp, pan,
          amplitude_envelope)
          midinote -= 1
               start += loop
          }

     }

//Day 89
rtsetparams(44100, 2)
load("STEREO")
load("MIX")
load("REV")
load("PAN")
load("MOOGVCF")
srand()
print_off()

rtinput(“/path/to/file.aif”)

bus_config("MIX", "in 0-1", "aux 0-1 out")
bus_config("REV", "aux 0-1 in", "aux 2-3 out")
bus_config("MOOGVCF", "aux 2-3 in", "aux 4-5 out")
```

```
bus_config("PAN", "aux 2-3 in", "out 0-1")

/*
Audio file is about 15 seconds long, so loop through it four times, or about 60 seconds
*/
totalduration = DUR() * 4

//bus_config("STEREO", "in 0-1", "aux 4-5 out")
STEREO(75, 0, DUR(), 3.0*maketable("line", 10, 0,0, 0.5,1.0, 1.0,0), 0.5)

////////////////////////////////////////////////FIRST GESTURE
//-----------------------------MIX
start = 0
instart = 0
duration = DUR()
amplitude = 1.0
envelope = maketable("line", 1000, 0,0, 0.5,1.0, 0.6,1.0, 0.75,0)
left = 0
right = 1

for(start = 0; start < DUR()* 12; start += DUR()){
     MIX(start, instart, duration, amplitude*envelope, left,right)
     }

//-----------------------------REV (just to warm up sound a bit)
start = 0
instart = 0
duration = 60
amplitude = 0.125
type = 1     // 1 is Perry Cook's, 2 is John Chowning's, 3 is Michael McNabb's
rvbtime = 30
rvbpct = 0.99999
inchan = 0

REV(start, instart, duration, amplitude, type, rvbtime, rvbpct, inchan)

//-----------------------------MOOGVCF
start = 0
instart = 0
duration = totalduration
amplitude = 0.75
inchan = 0
```

```
pan = maketable("random", 10, "even", 0,1)
bypass = 0
lowcf = 800
highcf = 10000
lowres = 0.9
highres = 0.25
centerfrequency = maketable("line", "nonorm", 1000, 0,lowcf, 0.3, highcf, 1.0, lowcf+200)
resonance = maketable("line", "nonorm", 1000, 0,lowres, 1.0, highres)
MOOGVCF(start, instart, duration, amplitude, inchan, pan, bypass, centerfrequency,
resonance)

//-----------------------------PAN
start = 0
instart = 0
duration = totalduration
amplitude = 0.75
inchannel = 0
pantype = 1
pan = 0.5
PAN(start, instart, duration, amplitude*envelope, inchannel, pantype, pan)

////////////////////////////////////////////////SECOND GESTURE (start at 30 seconds in)
load("PANECHO")
load("FREEVERB")

bus_config("PANECHO", "aux 0-1 in", "out 0-1") //from MIX above
bus_config("FREEVERB", "aux 4-5 in", "out 0-1")

//----------------------MIX on sort of a MEAPsoft-type segmentation with random instarts

loop = 1

for(start = 0; start < DUR()* 8; start += loop){
     randomnumber_x = irand(0,15)
     randomnumber_y = irand(0,15)

     if(randomnumber_x < randomnumber_y){
          duration = randomnumber_y / 2
          instart = randomnumber_y
          }
     else{
          duration = randomnumber_x / 2
```

```
            instart = randomnumber_y
            }

      MIX(start, instart, duration, amplitude*envelope, left,right)
      loop = irand(0,3)
      }

//-----------------------------PANCECHO for delay
start = 0
instart = 0
duration = totalduration
amplitude = maketable("line", 1000, 0,0, 0.4,1.0, 1.0,0.0)
leftchandelay = maketable("line", "nonorm", 10, 0,0.5, 1.0,3.0)
rightchandelay = maketable("line", "nonorm", 10, 0,0.125, 1.0,3.0)
feedback = 0.001
ringdownduration = 1.0
PANECHO(start, instart, duration, amplitude*envelope, leftchandelay, rightchandelay,
feedback,ringdownduration)

//-----------------------------FREEVERB
start = 0
instart = 0
duration = totalduration
amplitude = 1.0
room = maketable("line", 1000, 0,0.001, 1.0,0.9)
predelay = 0.05
ringdur = 5.0
damp = 70
dry = 0
wet = 1.0
width = 100
FREEVERB(start, instart, duration, amplitude, room, predelay, ringdur, damp, dry, wet,
width)


#Day 90
from rtcmix import *

rtsetparams(44100, 2)
load("WAVETABLE")

import time
```

```
import math
import random

divisor = pow(10,8)
x = time.clock()
y = time.time()  / divisor

def drange(start, stop, step):
     r = start
     while r < stop:
          yield r
          r+= step

#---------- WAVETABLE
start = 0
duration = 2.5
amplitude = 32768
frequency = 30
pan = 0.5
waveform = maketable("wave", 1000, "sawup")
loop = 0.125
for start in drange (0, 50, loop):
     duration = random.random() / 100
     pan = abs(math.sin(start))
     frequency = ((math.sin(y) * x) + y)
     WAVETABLE(start, duration, amplitude, frequency, pan, waveform)
     loop = random.random() * 2


#Day 91
# Here is the origingal 3n + 1 score, now in Python

from rtcmix import *

rtsetparams(44100, 2)
load("WAVETABLE")

import random
random.seed()  # seed random numbers, like srand() in MINC

def drange(start, stop, step):
```

```
     r = start
     while r < stop:
          yield r
          r+= step

#---------- WAVETABLE stuff, initializing p-fields
start = 0
duration = 1
amplitude = random.randint(5000, 20000)
envelope = maketable("line", 1000, 0,0, 0.1,1, 0.3,0.7, 0.7,0.7, 1.0,0)
pitch = 60

increment = 0.1515
n = trunc(random.randint(1000, 1000000))

for start in drange(0.0, 100.0, increment):       # Boundaries and step size
     if n%2 == 0:   # if n is even...
          n = trunc(n * 3 + 1)     # do a 3n + 1 operation
          transp = 1.0        # and transpose up and octave
          amplitude = random.randrange(10000, 20000)# louder amplitude values
          pan = 0
     elif n%2 == 1:      # if n is odd
          n = trunc(n / 2)    # divide n by 2
          transp = -1.0       # and transpose down an octave
          amplitude = random.randint(5000, 10000)   # quieter, and integers only
          pan = 1

     if n == 1:          # if n equals 1
          exit()    # stop

     if pitch > 128:
          pitch = pitch - 127

     pitch = cpsmidi(n)  # translate n to pitch in MIDI
     dB = dbamp(amplitude)
     constpowpan = boost(pan) / 2
     WAVETABLE(start, increment * 0.8, amplitude * envelope, pitch, constpowpan)
     print n

#Day 92
from rtcmix import *
```

```
rtsetparams(44100, 2)
load("MBANDEDWG")

import random
srand()

#------------  MBANDEDWG
start = 0
duration = 0.125
amplitude = 8000
pitch = cpsmidi(70)
strikeposition = 0.3
pluck = 0
maxvelocity = 0.5
instrument = 1
pressure = 0.0
resonance = 0.9
constant = 0.8
pan = makeLFO("sine", 0.2, 0.0,1.0)
x = random.randint(40,70)
cutoff = 5

def drange(start, stop, step):
     r = start
     while r < stop:
          yield r
          r+= step

for start in drange(0, 15, 0.125):
     rand_num = random.randint(0,10)

     if (rand_num < cutoff):
          x += 4 # Go up by major third

     elif (rand_num >= cutoff):
          x -= 3 # Go down by minor third

     pitch = cpsmidi(x)

     if (pitch < 32):
```

```
          pitch = pitch + x * 10

     MBANDEDWG(start, duration, amplitude, pitch, strikeposition, pluck, maxvelocity, in-
strument,
     pressure,resonance, constant, pan)

#Day 93
from rtcmix import *

rtsetparams(44100, 2)
load("WAVETABLE")

import random
random.seed()

#----------- WAVETABLE
start = 0
duration = 5
amplitude = 700
frequency = 200
pan = makeLFO("sine", "nointerp", 0.2, 0,1)
waveform = maketable("wave", "nonorm", "nointerp", 2000, "buzz")

def drange(start, stop, step):
     r = start
     while r < stop:
          yield r
          r+= step

increment = 1.4

for start in drange(0, 41, increment):
     randomtest = random.randint(0, 100)

     if randomtest < 80:
          duration = 3

     elif randomtest >= 80:
        duration = 10

     WAVETABLE(start, duration, amplitude, frequency, pan, waveform)
     WAVETABLE(start, duration, amplitude, frequency - abs(randomtest/50), pan, waveform)
```

```
#Day 94
from rtcmix import *  # create a list of pseudo-random numbers from a for loop

rtsetparams(44100, 2)
load("WAVETABLE")

import random
srand()

my_list = [] # declare list of elements, say MIDI notes
num_elements = 10
for i in range(num_elements + 1):
     my_list.append(random.randrange(50,101,1))

print my_list

for start in range(num_elements + 1):
     WAVETABLE(start, 1.25, 20000, cpsmidi(my_list[start]), random.random())

#Day 95
from rtcmix import *
import random
import math

rtsetparams(44100, 2)
load("WAVETABLE")

notes = []
num_notes = 12

for i in range(num_notes + 1):
     notes.append(random.uniform(60,62))  #use uniform for random floating pt nums

print notes[2]

def pitch_in_Hz(current_note):     # long-hand MIDI to frequency conversion
     octave = num_notes
     raw_conversion =  440 * (math.pow(2, ((current_note-69) / octave)))
     return round(raw_conversion, 0)

#-------- WAVETABLE
```

```
start = 0
duration = 0.25
amplitude = 20000
pan = random.random()
waveform = maketable("random", 18, "cauchy", -1,1)

def drange(start, stop, step):
     r = start
     while r < stop:
          yield r
          r+= step

loop = 1
for start in drange(0,num_notes,loop):
     x = random.random()
     WAVETABLE(start*x, duration, amplitude, pitch_in_Hz(notes[start]), pan, waveform)

#Day 96
from rtcmix import *

import math
import random

lowest_note = 45
highest_note = 96

rawnotes = []
num_notes = 4  # generating four mallet chords

for i in range(num_notes):
     rawnotes.append(random.uniform(lowest_note,highest_note))

note_gamut = [round(elem, 0) for elem in rawnotes]
print note_gamut

# now, make them ascending and descending, like SATB-ish
note_gamut_ascending = sorted(note_gamut)
note_gamut_descending = sorted(note_gamut, reverse = True)
print "gamut ascending: %s" % note_gamut_ascending
print "gamut descending: %s" % note_gamut_descending

#Day 97
```

```
from rtcmix import *

import math
import random
random.seed()
# ------- distributions
# random.uniform(low, high)
# random.triangular(low, high, mode)
# random.gauss(mu, sigma)
# SEVERAL more (lognormvariate, expovariate, betavariate

lowest_note = 45   # range of instrument, but need to be cognizant of range of each hand
highest_note = 96
mode = highest_note - lowest_note

rawnotes = []
num_notes = 4

for i in range(num_notes):
     rawnotes.append(random.triangular(lowest_note,highest_note, mode)) #focus toward mid-
dle
note_gamut = [round(elem, 0) for elem in rawnotes]
print note_gamut

note_gamut_ascending = sorted(note_gamut)
print "gamut ascending: %s" % note_gamut_ascending

#  to be safe, range of spread on each hand is one octave or less

s = note_gamut_ascending[3]
a = note_gamut_ascending[2]
t = note_gamut_ascending[1]
b = note_gamut_ascending[0]

if s - a > 12 or t - b > 12:  # this is quite slick, just writing "or"
     s = s - 12
     t = t - 12

gamut_in_range = [b, t, a, s]
print "gamut in spread range: %s" % gamut_in_range

#Day 98
```

```
from rtcmix import *
import math

trichordone = [1, 3, 7]
trichordtwo = [2, 10, 6]
hexachord = trichordone + trichordtwo

# ascending normal order
hexachord_no = sorted(hexachord)
print hexachord_no

# reversed
def reversed(input_list):
     return input_list[::-1]

print reversed(hexachord_no)

# palindrome
def palindrome(input_list):
     return input_list + input_list[::-1]

print palindrome(hexachord_no)

# palindrome elided
def palindrome_elided(input_list):
     return input_list + reversed(input_list)[1:]

print palindrome_elided(hexachord_no)

# transpose
def transpose(input_list, transposition_value):
     return [mod(element+transposition_value, 12) for element in input_list]

print transpose(hexachord_no, 5)

# rotate
def rotate(input_list, rotation):
     return input_list[rotation:] + input_list[:rotation]

print rotate(trichordone, 2)
```

```
#Day 99
# more LISP --> Python
from rtcmix import *
import random

# first LISP example, although abs( ) is intrinsic to Python
def absolute_value(x):
     if x >= 0:
          x = x
     elif x <= 0:
          return x * -1

x = -1
print absolute_value(x)

# exploring conditional tests
def conditional_example(y):
     if y > 5:
          return y * 10
     elif y <= 5:
          return y * 5

y = random.uniform(0,10)
print y
print conditional_example(y)

# fibonacci number, when nth number is given (1=0, 2=1, 3=1, 4=2, etc...)
def fibonacci(nth_num):
     if nth_num == 1:
               return 1
     elif nth_num == 0:
               return 0
     else:
               return fibonacci(nth_num - 1) + fibonacci(nth_num - 2)

print fibonacci(7) #seventh number in the fibonacci series
# construct a series, up to a given iteration
def fibonacci_series(iterations):
     a, b = 0, 1
     for _ in xrange(iterations):
               yield a
               a, b = b, a + b
```

```
print list(fibonacci_series(13)) # return thirteen numbers in the fibonacci series

#Day 100
from rtcmix import *
from random import randrange

rtsetparams(44100, 2)
load("WAVETABLE")

trichordone = []
trichordtwo = []
num_elements = 3
for i in range(num_elements):
     trichordone.append(randrange(10))
     trichordtwo.append(randrange(10))

     print trichordone
     print trichordtwo
     hexachord = trichordone + trichordtwo

def reversed(input_list):
     return input_list[::-1]

def palindrome_elided(input_list):
     return input_list + reversed(input_list)[1:]

def hexachord_to_midi(pitches, range):
     return [x + range for x in pitches]

pitches = hexachord_to_midi(palindrome_elided(hexachord), 60)  # 0 = C, 1 = C#,  etc..
print pitches

#------ WAVETABLE
start = 0
duration = 1.5
amplitude = 20000
envelope = maketable("line", 1000, 0,0, 0.3,1.0, 0.5,0.5, 0.75,0.5, 1.0,0)
pitch = cpsmidi(pitches[start])
pan = 0.5

for start in range(0,12):
```

```
pitch = cpsmidi(pitches[start])
WAVETABLE(start, duration, amplitude*envelope, pitch, pan)
```

|| Useful links in the world of RTcmix ||

What follows are a list of helpful links that I hope you'll seek out as you continue your work with RTcmix.

1. RTcmix website - http://rtcmix.org

   If you're like me, you'll probably be referencing this site often. In fact, when I'm working in our main studio at Crane with dual monitors, I'll have the RTcmix website open on one screen and my Terminal window on another. Its "Reference" section is an invaluable resource for fully understanding the ins and outs of each instrument or command at hand. Moreover, this will point you in the direction of some useful tutorials and links to music, artist pages, and examples from users in the RTcmix community.

2. RTcmix GitHub page - https://github.com/RTcmix/RTcmix

   The transition to hosting RTcmix on GitHub is a relatively recent occurrence, in that it was completed in the aforementioned four years that went into this text. A huge advantage of GitHub is the ability to test our experimental releases of the program, called "branches."

3. RTcmix mailing list - http://music.columbia.edu/cmc/RTcmix/

This is by far the best way to keep up to date on all things RTcmix. The original authors are still highly active and are never more than an email away from answering any questions that you might have, though I of course hope that many of your questions regarding downloading, installing, implementing, etc have been covered in some way. I'm on the mailing list, too, so you'll find me in the conversation as well. There is a separate mailing list for RTcmix developers, which really should be reserved for build questions, architectural inquiries, and potential "bugs" and fixes.

4. Soundcloud page - https://soundcloud.com/groups/rtcmix

Your one stop shop for music made with RTcmix! This is a motivated, active group that cares deeply about and is interested in your music. Any new works that are created with RTcmix are welcome to be hosted on the group page and we would of course love to see that community grow, so please join us!

A number of works were mentioned throughout this text and I'd like to formally list them here, along with other sources that were used for the research of this book.

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All of the score files utilized in this text are hosted on my website. You can find them on the "RTcmix book" page. Moreover, I completed 365 daily score files in 2013 as part of my *Script Calendar* project. Those too are found on my website under "Works --> Electronic."

Please feel free to contact me with any questions you might have about the book, helpful suggestions for future edits, or anything else RTcmix related! I can be found on Twitter @pareidoliaudio.