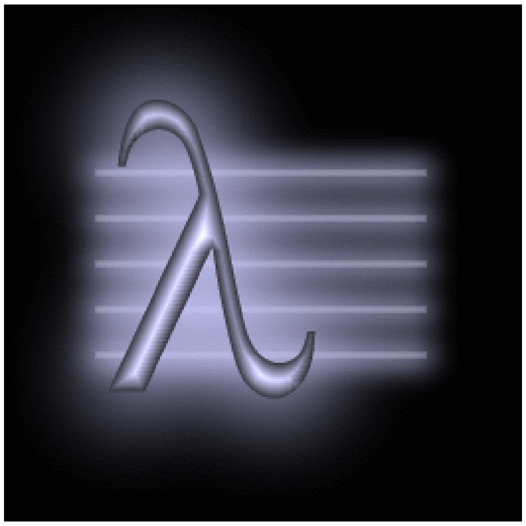
# Notes from the Metalevel

## An Introduction to Computer Composition



Heinrich Konrad Taube

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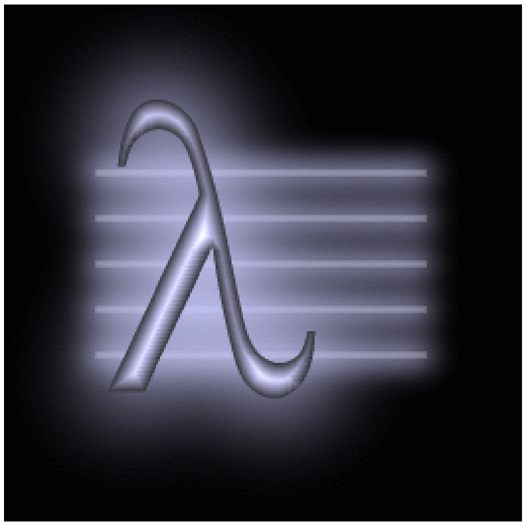

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# Preface

Notes From the Metalevel is a practical introduction to computer composition. It is primarily intended for student composers interested in learning how *computation* can provide them with a new paradigm for musical composition. Notes from the Metalevel explains through a practical, example-based approach the essential concepts and techniques of computer-based composition and demonstrates how these techniques can be integrated into the composer's own creative work. One of most exciting aspects of computer-based composition is that it is an essentially *empirical* activity that does not require years of formal music theory training to understand. For this reason, Notes from the Metalevel should also be of interest to any reader with a high-school mathematics background interested in experimenting with music composition using MIDI and audio synthesis programs. The book should also be of use to computer science and engineering students who are interested in artistic applications of object-oriented programming techniques and music software design.

Every composer knows that learning how to write music takes an enormous amount of practice. It is only through practice that new techniques can become integrated into the composer's own approach to the craft. That is why this book takes a “hands on” approach to its subject matter, by combining a general theoretical discussion with real software examples. Notes from the Metalevel is filled with examples and exercises for the student to study, modify and adapt for their own musical purposes. All of the examples as well as the software to work with them are available on the CD that accompanies the book. The CD also contains an HTML version of the book with active links to the program examples and the sound they generate. The reader is strongly encouraged to experiment with the chapter examples interactively, in parallel with reading the text. The vast majority of these examples are short, simple, and demonstrate only one or two points at a time so that the student may immediately begin the process of modification and experimentation. In addition to the many chapter examples, Notes from the Metalevel also contains a number of larger projects, called *Etudes*, that appear at regular intervals throughout the book. The Etudes are large structured examples that explore some technique or topic in greater detail.

## Plan of the Book

Notes From the Metalevel is divided into two parts. The first part presents an introduction to music programming in Lisp. It begins with a brief history of Lisp and discusses why the language is a particularly good choice for music composition. The rest of the chapters in part one give a *selective* introduction to computer programming techniques. The term selective means that the material in these chapters is specifically chosen for its relevance to musical programming and to the material presented in the second part of the book. One of the really wonderful benefits of working in Lisp is that a person does not have to be an expert in order to do meaningful and important work in the language. However, some composers who start working with computer languages want to develop technical expertise in parallel with their artistic pursuits. This book contain pointers to a wide variety of Lisp documentation freely available on the Web, including several famous Lisp text books, the Lisp and Scheme language specifications, and on-line Lisp tutorials. Students who want to learn more about Lisp are strongly encouraged to take the additional time necessary to read and study this reference material in detail.

The second part of the book is an introduction to the essential concepts and techniques of computer composition. The first few chapters are related to the representation of sound, musical structure, algorithms and processes. The material in the rest of the book can be broadly grouped into four areas of discussion: algorithmic design, mapping and transformations, aleatoric composition and pattern based composition. All of the examples and exercises use MIDI to produce musical score output. This allows the book to concentrate on compositional techniques rather than sound synthesis techniques. However, it is very easy to work with the examples using a *synthesis language* like CSound or Common Lisp Music. The last chapter in the book (Beyond MIDI) gives an overview of how to substitute synthesis languages in place of MIDI.

## Publication Formats

Notes from the Metalevel is published in two forms, the hard-copy version and an “on-line” HTML version that is available on the accompanying CD. The HTML version of the book uses Cascading Style Sheets to format its presentation and will look best in a browser that complies fully with the CSS2 and XHTML standards.

## Typographic Conventions

Both versions of the book follow special typographic conventions to make the presentation as consistent and understandable as possible.

### Programming Examples and Lisp Interactions

Notes from the Metalevel contains two types of examples: programming examples and Lisp interactions. The text for both type of examples always appears in lower case letters in a fixed width font. Programming examples consist of Lisp expressions that would typically be edited in text files and then evaluated in Lisp. These examples appear between dashed upper and lower borders to indicate that the portion displayed is a part of a larger file. Programming code is displayed using *syntax highlighting* to stress important syntactic units. The hard copy book and the XHTML book differ in how this highlighting is depicted. In the XHTML version the highlighting is displayed as colored text. Normal code appears in black face, definitions and *special forms* (forms like **`define`** and **`if`**) appear dark red, user comments are blue, text strings are green and keyword symbols (symbols starting with a colon) are red. In the hard copy version of the book, Lisp comments appear in italic face, and the other important units (definitions, special forms and keywords) appear in bold face (Example 1).

Example 1. Example program code with syntax highlighting.

```
(define (fibonacci nth)
  ;; return the nth fibonacci number
  (if (<= 0 nth 1)
    1
    (+ (fibonacci (- nth 1))
       (fibonacci (- nth 2)))))
```

Lisp interactions consist of expressions that the user will type in the Listener as commands for Lisp to execute. Interactions are shown inside solid bordered boxes with a light grey background to indicate that the contents is something that the user is typing inside the *Lisp Listener*, a window that supports an interactive command shell in Lisp. Within the Listener window the *Lisp input prompt* **`cm>`** marks the beginning of every input line and the user's input expressions appears in bold face just after the prompt. Output that results from Lisp's evaluation of the user's input expression appears in regular face on the next line. If no output was produced then the next line is left blank. More than one Lisp interaction may appear within a single interaction box, in which case the more recent interactions appear toward the bottom of the example. The last line of an interaction shows the prompt line with no input to indicate that the Lisp Listener is waiting for the user to type the next input expression (Interaction 1).

Interaction 1. Example interactions in the Lisp Listener.

```
cm> (list 1 2 3)
(1 2 3)
cm> (define 2pi (* 2 pi))

cm> 2pi
6.283185307179586
cm>
```

Within a given chapter, each programming and interaction example is numbered. Examples usually depend on the sequential evaluation of all preceding examples and interactions in the chapter.

### Chapter Source Text

The best way to read the book is to run the Common Music application in parallel as you study each chapter. This will allow you to test out each interaction and experiment with the program examples as you encounter them in the text. The source code for all of the program examples and interactions can be found in a ".cm" file (Common Music source file) located in the same directory as the HTML file for the chapter. This source file can be edited and evaluated inside the Common Music application. Etudes, op. 1 describes two different ways to work with example material.

## Sound Examples

Almost every musical example in the book has one or more sound examples associated with it. These files contain the musical output generated by the example. Audio files are stored in stereo MP3 format and MIDI files are stored in Level 0 (single track) format. Except for a few noted cases the MIDI files do not contain any program changes. This allows the reader to experiment with different instrumentation while listening to the examples. Each sound example is marked in the text by an arrow. Sound examples in the hard-copy book contains the location of the sound file in the chapter directory. Sound examples in the XHTML document are links that play the example. A short description of the contents of the file may appear with the sound example. If no instrumentation for a MIDI file is suggested then a Grand Piano sound may be assumed. All the examples have been tested in Quick Time version 6.0. The following sample MIDI and MP3 files can be used to test your audio setup:

- sines.midi Two midi "sine waves" at slightly different rates.
- thunder.mp3 Tuned thunder clap from Aeolian Harp.

## Special Notes

Special notes highlight points of difference between Lisp implementations or operating systems that affect how the examples run or appear on the various different systems. A special note consists of a styled label and an explanatory text:

CLTL Note:
Common Lisp defines `nil` to mean both boolean false and the empty set.

## Dictionary Terms

Functions and variables defined by Common Music are documented using a special typographic format. Definition entries, literals (code that you type) and program code are printed in fixed face. Variables and metavariables (names that stand for pieces of code) are printed in italic face. Terms separated by | are exclusive choices. Brackets {} and square brackets [] are sometimes used to associate or group the enclosed terms in a syntax declaration. These brackets are never written in actual code. Square brackets [] mean that the enclosed terms are optional. An optional term may appear at most one time. Curly brackets {} simply group the enclosed terms. A star * after curly brackets means the enclosed terms may be specified zero or more times. A plus + after curly brackets means the bracketed term may be specified one or more times. A sample syntax definition:

[Function]
`(`**`hertz`** *x* `[:in` *scale*`])`

In the sample above **`hertz`** is the name of the function, *x* is a required argument and *scale* is an optional keyword argument. Italicized variables in a definition may have descriptive names or may consist of a single letter such as *x*. Single letter names generally provide an indication as to the type of data the user is expected to specify. The meaning of the letter variables are shown in Table 1.

Table 1. Single letter variable names.

| **Letter** | **Description** | **Letter** | **Description** |
|---|---|---|---|
| *x* | any value | *b* | boolean |
| *n* | number | *s* | symbol or keyword |
| *f* | float | *l* | list |
| *i* | integer | *g* | string |

Dictionary terms are only briefly explained in the main chapter text. The full explanation of these entries can be found in the on-line Common Music Dictionary. Terms in the HTML version of the book provide a link to the main dictionary entry.

## Acknowledgments

I would like to thank the Research Board at the University of Illinois for its grant toward the completion of this manuscript. The author is indebted to the Center for Computer Research and Acoustics (CCRMA) at Stanford University and to the Zentrum für Kunst und Medientechnologie (ZKM) in Karlsruhe, Germany for their contributions to the Common Music software project on which this book is based. I would like to thank Johannes Goebel, director of the Audio group while I was working at ZKM, for his commitment to the Common Music project while I was employed at the center. The author would like to acknowledge William Schottstaedt at CCRMA for his contributions to computer music programming in general and for his Common Lisp Music (CLM) synthesis language which the author used to produce the Aeolian Harp examples included on the CD. The author is greatly indebted to Tobias Kunze for his many contributions to Common Music and for his help in the preparation of this document. Tobias implemented many of the MIDI features in Common Music, wrote several examples on which Etudes chapters in this book are based, designed Common Music's glowing lambda logo and developed the CSS style for the HTML version of the book. I would also like to express my thanks to Richard Karpen and the Center for Advanced Research Technology in the Arts and Humanities (CARTAH) at the University of Washington for their friendly encouragement and for the many good improvements and suggestions that they have made over the years. And finally, I would like to thank Carl Edwards and David Phillips for their careful reading of this manuscript and for their sound editorial advice.

## Colophon

This book was written in the Xemacs text editor using the ispell spelling program and a lisp-based mark-up language called Mtxt implemented by the author. The Mtxt mark-up language generated the XHTML version of the book as well as the chapter example files. The XHTML output from Mtxt was then imported into Microsoft Word with a different CSS style sheet to produce the camera-ready manuscript. All of the graphic plots that appear in the book were created in Plotter, a plotting tool included in the Macintosh/OS 9.2 version of Common Music. The illustrations were designed in AppleWorks 6. A portable version of Plotter that runs in all versions of Common Music is currently under development. Both the Mtxt implementation file as well as Plotter code can be found on the CD that accompanies this book.

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H. Taube
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*to my children*

*Konrad Gabriel, Maralena Valentin, and Florian Alexander Taube*

There be none of beauty's daughters
With a magic like thee;
And like music on the waters
Is thy sweet voice to me.

— Lord Byron, Stanzas for Music

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# 1 Introduction

The one crowded space in Father Perry's house was his bookshelves. I gradually came to understand that the marks on the pages were trapped words. Anyone could learn to decipher the symbols and turn the trapped words loose again into speech. The ink of the print trapped the thoughts; they could no more get away than a doomboo could get out of a pit. When the full realization of what this meant flooded over me, I experienced the same thrill and amazement as when I had my first glimpse of the bright lights of Konakry. I shivered with the intensity of my desire to learn to do this wondrous thing myself.

spoken by Prince Modupe, a west African prince who learned to read as an adult.

— Leonard Schlain The Alphabet Versus the Goddess

It is impossible to know exactly how Prince Modupe felt when he discovered a process by which his very thoughts could be trapped and released at will again into speech. But I think his epiphany must be close to what I experienced when, as a young composer, I was first shown how I could use a computer to represent my musical ideas and then “release them” into musical compositions. At that time I was a masters student in composition at Stanford University. My teachers John Chowning and Leland Smith had just demonstrated to our introductory computer music class how a music language called SCORE could capture musical ideas and, at the push of a button, trigger an almost magical process in which fantastically complex scores were computed and then realized by instruments unimpeded by the laws of physics. At that instant it became clear to me that there was an entire level of notation above the scores that I had been writing in my regular composition classes, a level I knew nothing about! But I could see that in this level it was possible to notate my compositional ideas in a precise manner and work with them in an almost physical way, as “trapped words” that could be unleashed into musical sound whenever I wanted. Equally important was the realization that this new (to me) level was essentially devoid of any preconceived notions of “musical correctness” and so offered me an attractive alternative to the symbols and glyphs of Common Practice music, which I increasing felt were too tied to historical tradition. I too, “shivered with a desire to do this wondrous thing myself” and, as I look back on it now more than twenty years later, I realize that this moment was one in which my life's path was irrevocably altered.

The first several years that I was associated with the Center for Computer Research in Music and Acoustics (CCRMA) were among the most intellectually stimulating experiences I have ever had. Our computer music classes were located at the D. C. Power Laboratory (named after a person, not a current), an off-campus facility that was devoted primarily to research in Artificial Intelligence. The Lab housed a PDP-10 mainframe computer that students in the computer music group were allowed to use during the night from 2-7am. As I learned more about music languages during my studies I also took time to learn about the software languages that AI researchers were using at the Lab. I cannot say with certainty why I did this since I had never taken a class in computer science and was not particularly good at mathematics! There were two primary languages in use: SAIL (Stanford Artificial Intelligence Language) and LISP, a language developed by James McCarthy also at Stanford. I was immediately attracted to the power and beauty of the Lisp language (for many of the reasons that I outline in the next chapter of this book) and I began to use it in my musical life.

All things change, particularly in the digital world, and one day the SAMBOX digital synthesizer that was at the center of my artistic life was retired without any immediate replacement. My software world — which was hopelessly tied to the retired hardware — vanished at the speed of an electron, and all of the programs and libraries I had used for years were suddenly useless. I decided at that point that I would develop a generalized and portable replacement to that world, one that would run on any machine, and could be customized and redefined even to the core of the language. The result was the Common Music software system, written in Lisp, that continues to evolve today. Though born of necessity, Common Music developed in response to the explosive growth of new hardware and music systems that resulted from the introduction of the personal computer in the 1980's. As the number of computers, operating systems, sound synthesis languages and music applications increased, the need for a portable composition language that could function with all of these systems became obvious. The Common Music software project started in 1989 when I was a Rockefeller Fellow at CCRMA. Much of the system as it exists today was implemented at the Institut für Musik und Akustik at the Zentrum für Kunst und Medientechnologie in Karlsruhe, Germany, where the author worked for the first half of the 1990's and in 1996 Common Music received First Prize in the computer-assisted composition category at the 1er Concours International de Logiciels Musicaux in Bourges, France. Common Music continues to evolve today at the University of Illinois at Urbana-Champaign, where the author is now an associate professor of music composition.

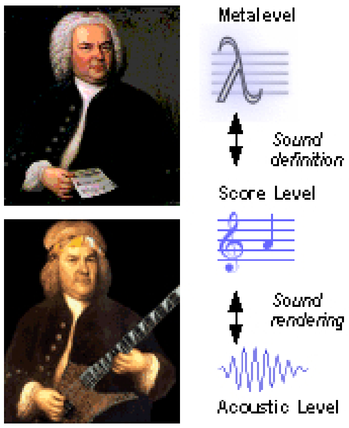
## Notes from the Metalevel

Every musician is familiar with performing music from a score. The score is not the same thing as the music, it is a *representation* of music — a notation designed to capture information about creating sound with an instrument. Musical sound can be represented in a number of different ways and on a number of different levels. For example, a recording represents music at a very low level, by storing a representation of the music's acoustic waveform. Of course, the recording does not contain any performance information. Musical notes and other kinds of performance instructions can be thought of as constituting a *level of abstraction* above the sonic waveform.

If we compare the score of a composition with its recording there are several observations that can be made. First, the score is a much more condensed representation than the recording. By definition, the process of abstraction eliminates, or hides, a certain amount of detail that exists in the unabstracted layer. In addition to hiding detail, an abstraction might also be a more *general* representation than the information in the unabstracted layer. Another way of saying this is that, while the abstraction is more concise it may also be less exact. Imprecision does not mean that something is wrong with the abstraction, just that there is not an exact correspondence between it and the lower level. For example, given a musical score a performer can create different acoustic level representations simply by performing the score a number of different times. In fact, it is the imprecision of traditional scores that enable the various musical interpretations that make performance art interesting.

A score such as the one Bach is holding in Figure 1 is written in a language that allows a composer to convey information to a performer. The language provides a set of *symbols* — the musical marks that appear on the page — and a *syntax*, or grammar, for stringing the symbols together to form meaningful statements. Given the existence of the acoustic and score representations one might ask if there is yet another representation that constitutes a level of abstraction above the performance score? The answer, of course, is yes; it is what this book terms the *metalevel*. If the score represents the composition then the metalevel represents the *composition of the composition.* A metalevel representation of music is concerned with representing the activity, or *process*, of musical composition as opposed to its artifact, or score. If the metalevel seems more ephemeral than the performance level it may be due to the fact that it is more closely related to the mysterious cognitive processes that occur within the composer. In addition, there is no physical artifact, such as a score or a CD, to represent it. This book is about using the computer to *instantiate* this level: to define, model and represent the compositional processes, formalism and structures that are articulated in a musical score and acoustic performance but are not literally represented there. By using a computer the composer can work with an explicit metalevel notation, or language, that makes the metalevel as tangible as the performance and acoustic levels. But unlike the bits on a CD or the notes on a score, statements in the metalevel representation are active agents for the processes, methods, algorithms and techniques that a composer develops to craft the sounds in his or her compositions.

Figure 1-1. Music can be represented on different levels of abstraction, acoustic, performance and metalevel. (The metalevel is depicted using the Greek lambda superimposed on a music staff.



## Computer Composition

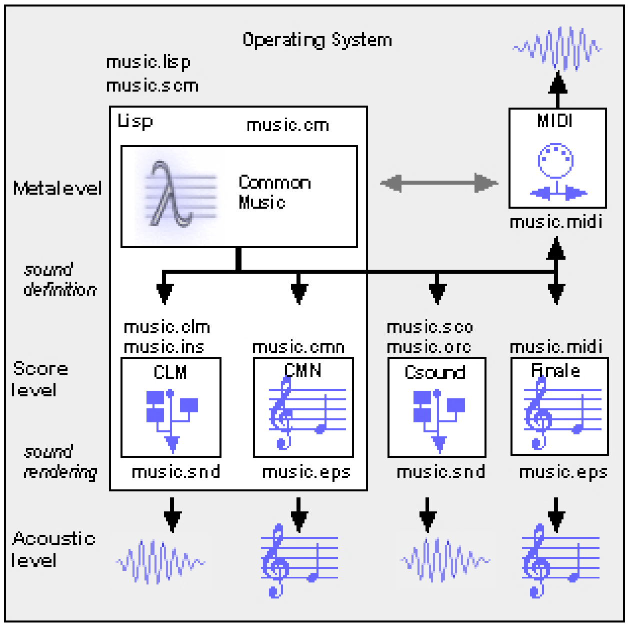
Over the course of the last few decades the computer has played an ever increasing role in the professional lives of composers. Computers are now involved in all phases of a composer's work, from pre-composition exploration to sound generation, score notation and CD recording (Figure 2). Composers have been using the computer as a compositional tool for almost as long as the modern computer has been in existence. Lejaren Hiller and Leonard Isaacson are generally regarded to be the first “computer composers” and their *Illiac Suite* for string quartet was premiered August 9, 1956 at the University of Illinois at Urbana-Champaign, where the present author is currently on the composition faculty. Over the course of the last few decades the computer has come to take on new roles in music composition. It is now useful to distinguish between several different ways computers can applied to the specific activities in music composition. In *computer assisted composition* the computer is used to help with compositional tasks: generating pre-compositional data, facilitating playback and editing, and so on. In some sense the computer is used before and after the composer has an idea and is not really involved with modeling the composition itself. The term *automatic composition* can be applied to computer systems that are designed to compose music independently. Software such as David Cope's Experiments in Musical Intelligence and certain types of sound installations are examples of this kind of system. *Computer-based composition* means to use the computer to explicitly represent compositional ideas at a higher level than the performance score. An explicit metalevel representation means that the relationships and processes that constitute a composition (the composition of the composition) exist inside the machine apart from the composer thinking about them.

Why would a composer be interested in computer composition? The point of composing with a computer is to think about and to write music differently than one would without using a computer, by representing compositional formalisms in an explicit, transformable and computable manner. This affects the compositional process as well as the composition! These effects can be seen in a number of different ways:

- Composition becomes an *empirical* activity, one in which the composer experiments with ideas and rapidly tests them out as they develop from an initial curiosity into their final compositional form. Regardless of whether or not a composer writes “experimental music”, computer-based composition practically insures that the composer writes “experimenting music”.
- An explicit metalevel representation allows compositional ideas to be treated as modular building blocks that may be useful in a variety of musical contexts and in different musical pieces.
- Compositional tasks can become generic solutions organized in “toolboxes” that make it easier to incrementally develop compositions.
- Just as it is possible to generate different acoustic representations from a single score, so it is possible to generate different versions of a score from the same metalevel representation.
- Complex probability and chance procedures can be incorporated into the compositional process very easily.
- Very small changes at the metalevel can have an enormous impact on the score level representation.

- Relationships simply expressed in a metalevel notation can generate extremely complex scores.
- Changes in the metalevel can lead to unintended or unanticipated consequences that trigger new compositional ideas or take the compositional process in a totally new direction. Any composer who has integrated computers their compositional process in an experimental way can cite examples of this phenomena in their own music.
- An explicit metalevel representation means that compositional ideas can be manipulated and processed by machines. If the composer can construct a metalevel representation, then so can a program. This means the metalevel (the composition of the composition) could be an artifact of other, higher order levels of abstraction in the machine.
- Metalevel compositions can explore domains too complex or too large to imagine doing by hand. Of course, since a computer is deterministic then strictly speaking there is nothing that can be accomplished with a computer that cannot also be accomplished with a pen and paper. But the quantitative difference in speed between computer-based and pen-and-paper-based computation is of such a magnitude that for all intents and purposes the quantitative distinction is a qualitative difference for the metalevel composer.

Figure 1-2. The composition abstraction levels represented in a computer and their relation to the software systems described in this book.



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# 2 The Language of Lisp

Just as its hard to describe what makes a piece of music or a painting aesthetically pleasing, it's equally difficult to describe what makes a mathematical theorem or a physical theory beautiful. A beautiful theory will be simple, compact, and spare; it will give a sense of completeness and often an eerie sense of symmetry.

— Charles Seife, Zero, The Biography of a Dangerous Idea

## What is Lisp?

Lisp is a computer programming language written in Lisp. Behind this tautology lies one of the most unique and interesting facts about the language, namely, that Lisp does not make a distinction between data, the grist of computation, and programs, the software agents that act on data. A Lisp program *is* Lisp data. This symmetry between programs and data reflects a very different "world view" than computer languages like C or Fortran. What are some of the consequences of Lisp's unique approach to language design that are relevant to the activity of music composition?

### Lisp is easy to learn

Lisp's syntax is simple, compact and spare. There are only a handful of "rules" to know about. This is why Lisp is sometimes taught as the first programming language in university-level computer science courses. For the composer it means that useful work can begin almost immediately, before the composer understands much about the underlying mechanics of Lisp or the art of programming in general. In Lisp one learns by doing and experimenting, just as in music composition.

### Lisp is beautiful

It might seem odd to call a programming language beautiful but many people who use Lisp are attracted to it as much by its elegance as by its efficacy. Like the M. C. Escher hand that draws itself, the recursive, self-similar nature of Lisp can be inspiring. The symmetry between data and programs gives the whole language a sense of circularity and completeness that makes working in Lisp, for many people, as much an aesthetic experience as it is a practical one. This is, I think, an important quality for the novice Lisp composer to be aware of and one that composers are particularly well suited to appreciate. Aesthetics can play just as important a role at the compositional metalevel as they do at the performance level.

### Lisp is interactive and dynamic

When a Lisp application runs, a window called the Listener is available for the composer to interact with the Lisp environment. This interaction allows the composer to rapidly develop new ideas and test them out “on the fly”. Lisp's interactive nature also encourages the composer to start developing a musical program even before fully understanding how it will work or what its final outcome will be. This style of programming is called prototyping. In prototyping, an application is developed incrementally, through a process of experimentation and change. By its very nature Lisp supports the exploratory and incremental process of music composition. Lisp allows the composer to start out with an initial idea and then gradually work out the composition through an organic process process of development. This natural, “bottom up” style of working is much more difficult to do in other computer languages. Most languages force the composer to adhere to a rigid top-down “edit-compile-link-execute-test” process. Changing one line of code in one procedure forces the entire process to start over. Lisp's bottom-up style avoids rigidity in the development process — Lisp programs can be compiled or not, altering a function definition requires only that the definition be reevaluated. The composer can interactively test a part of a music program in isolation before adding it back into the general musical context.

### Lisp is a programmable language

The fact that Lisp is written in Lisp means that the language itself can be programmed. Lisp's programmable nature makes it an ideal metalanguage for implementing other languages and large software systems in. Lisp naturally encourages levels of abstraction to be implemented in software. The abstraction levels in computer music — composition, synthesis and sound production — can be reflected by different Lisp software layers all of which work together to further a common goal and are in fact indistinguishable from the underlying language itself.

### Lisp makes symbolic computation possible

All programming languages calculate with numbers but very few languages are designed from the bottom up to process symbolic expressions. A symbolic expression is an expression that consists of words and groups of words. Lisp programs can manipulate symbolic expressions to implement language processing. Music is a symbolic language and grouping is fundamental to music composition. The ability to work with symbols and groups of symbols in Lisp makes it very easy to represent and manipulate musical structure.

### Lisp conforms to a language standard

The language standard insures that Lisp operates the same way on all of the different computers and operating systems currently available today. There are many free implementations of Lisp that adhere to this standard, which means that the Lisp composer can be productive on practically any computer available today anywhere in the world at no expense.

## The History of Lisp

Lisp is one of the oldest computer programming languages. It was invented by John McCarthy in the 1950's to support his research into symbolic computation and artificial intelligence (AI). As a simple demonstration of what symbolic expressions are and what a Lisp AI program might do with them, consider the following two expressions:

1. Bach is a composer
2. All composers are mortal

A program that performs pattern matching and inference can conclude a new fact, namely:

3. Bach is a mortal

This is interesting, in part, because if you read the first two sentences (or listen to Bach's music!) you wouldn't necessarily conclude that Bach is a mortal. But by applying generic inference rules to a database of expressions the program can add new expressions to its knowledge base. In some sense, then, such a program has “learned” since the “knowledge” at its disposal has measurably increased.

Although research in Artificial Intelligence was one of the initial reasons for developing Lisp, it has long since ceased to be the central focus of the language. Over the last few decades Lisp has evolved into a general purpose programming language that is used for many different types of applications: CAD systems, notation programs, text editors, HTML servers, window managers, even airplane reservation systems. But at its core, Lisp remains a language that is particularly well suited to research and, by extension, to creative domains such as music composition that have a strong exploratory nature.

## Dialects and Implementations

As research into symbolic computing progressed in the 1960s and 1970s a number of different Lisp dialects developed. A *dialect* is a version of a language. Dialects arise, in part, because different groups of researchers may, over time, develop different or specialized requirements that must be implemented by modifications to an underlying, common language.

Scheme and Common Lisp (CL) are the two most important Lisp dialects in existence today are. Each dialect has its particular strengths: Scheme is a small, elegant, consistent and pure version of Lisp; Common Lisp is a much larger, generally faster and more “industrial strength” version of Lisp.

Within a single Lisp dialect there are likely to be a number of different *implementations*. An implementation is the computer software that makes a specific instance of a language. Some Lisp implementations are free, others are products sold by companies. In general, all implementations of a given dialect agree on a common set of features as defined by a *language specification*. A language specification is a document that defines the exact syntax and semantics of the language. The specification for Common Lisp is Common Lisp: the Language (2nd Edition), by Guy Steele. The specification for Scheme is The Revised (5) Report on the Algorithmic Language Scheme.

Lisp implementors are free to add extensions to the core language defined in the language specification. Features not explicitly defined by the specification tend to vary from implementation to implementation. Common extensions include “development tools” such as compilers, graphics libraries, specialized support for operating systems, program code analyzers, optimizers, program debuggers and so on.

## Common Music and Lisp

The music software system that accompanies this book is called Common Music (CM). Common Music produces sound by transforming a metalevel representation of music into a variety of different protocols for controlling sound production and display. The system was originally written in Common Lisp but now includes both Scheme and Common Lisp bindings (Table 1). Because Scheme is smaller and conceptually more consistent than Common Lisp it is the dialect that is used and documented in Notes from the Metalevel, and the examples shown in the book appear as they would look in Guile 1.6.3 Scheme running on the OSX operating system.

There are several things to keep in mind while working through the book. First, although the book adopts Scheme syntax for pedagogical reasons all of the examples and program code in Notes from the Metalevel will work with *any* implementation of Common Music that is listed in Table 1 because the Scheme functions and variables documented in this book are available in CM running in the Common Lisp implementations. Secondly, any extensition provided by the host Lisp implementation are available to the user when working with Common Music, but Common Music itself does not depend on them nor require them. And lastly, the vast majority of examples in the book do not depend on any operating system specific features. Where such differences do exist they are documented using one of the special notes discussed in the Preface.

Composers who would like to examine CM's program code to learn how a particular feature or function was implemented can access the source code on the accompanying CD. Common Music's Scheme sources have a .scm extensions and Common Lisp sources have a .lisp extension. The sources included on the CD are a "snapshot" of the software as it was at the time this book went to press. Common Music continues to evolve and the reader may wish at some point to install a newer version of the sytem than the one that was shipped on the book's CD. Source code to the most recent version of Common Music can always be downloaded via the Internet from the CM project homepage.

Table 2-1. Lisp Implementations for Common Music

| **Implementation** | **Dialect** | **Linux** | **OSX** | **MacOS** | **Windows** |
|---|---|---|---|---|---|
| ACL | CL | x |  | x | x |
| CLISP | CL | x |  | x | x |
| CMUCL | CL | x |  |  |  |
| Guile | Scheme | x | x |  | x |
| MCL | CL |  | x | x |  |
| OpenMCL | CL | x | x |  |  |

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# 3 Functions and Data

Lisp is a *functional language*, which means that all actions are accomplished by functions acting on data. A Lisp function is a program, or procedure, that accomplishes a task. The term *data* refers to the information that a function acts on, or processes. The use of functions in Lisp comes from the Lambda Calculus, a branch of mathematical logic developed by Alonzo Church in the 1930s. The Lambda Calculus defines mathematical computation in terms of anonymous functions, called *lambda expressions*, that are applied to input values to produce output values. This basic computational model is adopted by Lisp but with two important additions. First, most Lisp functions are not "anonymous", they have names associated with them. A name makes it easy to refer to a function and also gives an indication of what purpose the function serves. Secondly, not all functions in Lisp return results. Most Lisp functions behave like lambda expressions — they calculate useful values. But some functions are used for the *effects* they produce rather than for a value they return. For example, the purpose of the Lisp function **`define`** is to add a new definition to the language rather than to compute a result.

Interaction 3-1. Examples of calling Lisp functions.

```
cm> (* 2 pi)
6.283185307179586
cm> (define 2pi (* 2 pi))

cm> (list 2pi 2pi)
(6.283185307179586 6.283185307179586)
cm>
```

In Interaction 3-1 the function **`*`** returned the product of its inputs 2 and *pi*. In the next line of input the function **`define`** added a new variable definition to Lisp and did not return a value. The terms *function, lambda body, procedure, special function, macro,* and *closure* are used at various times in this book to refer to types of Lisp functions. Two of these terms deserve mention here because they will be used in the next several chapters of this book. A *special function* is a basic building block of the Lisp language. Special functions are "special" because they have their own unique syntax that provides some essential feature of the Lisp language. The **`define`** function is one such special function. A *macro* is a function that rewrites expressions. A macro takes an input expression exactly as the user typed it and then *transforms* it into a different expression which Lisp then executes. This is a subtle but important point because it means that the notation of the external expression may follow its own special rules of grammar. *loop* and **`process`** are two very important macros in this book that appear in almost all the music composition programs presented after Chapter 9.

## Data and Datatypes

The term *data* refers to the information that a function acts on, or processes. Most functions receive input data, perform a computation, and return an output value back to the caller. In some sense, then, a function can simply be thought of as an object that maps input values to output values:

Table 3-1. Functions map inputs to outputs.

```
 Inputs:    1   2       1  2  3           60
             \ /         \ | /            |
Function:     +          list      keynum->hertz
              |            |              |
 Outputs:     3         (1 2 3)        261.625
```

Since we only interact with functions by passing inputs and receiving outputs we do not need to understand how they are implemented in order to use them. For the moment we will just treat them as “black boxes” that provide useful programming services.

One of the most important skills that the beginning music programmer can develop is an ability to understand what sorts of information will be needed for given musical project and how best to represent that information in the actual data that a Lisp program will process. The composition programs and algorithms that appear in this book process many different kinds of information, from simple numbers to complex and multifaceted objects in object-oriented programming. One of the basic features of the Lisp language is that all data are classified according to their *type*. Consider a program that converts a MIDI key number to a Hertz frequency value. If key number 60 (Middle C) is the input the program would output 261.625 Hertz. While both the input and output are numbers they are actually different *types* of numbers. The number 60 is an exact number called an integer. The value 261.625 is an inexact, or real number. The terms integer, real and number are examples of Lisp *datatypes* that classify the data that Lisp programs process. The novice Lisp composer must develop an ability to recognize the most common types of data used in Lisp programming, to understand how these datatypes can be used to represent musical information, and to be able to specify this data in proper Lisp notation.

## Symbolic Expressions

The Lisp language was invented to support *symbolic computation*, a type of programming in which tokens, called symbols, can be processed along with quantitative information. A symbol can be thought of as a word. For example, the symbol **`debussy`** denotes a great French composer. In symbolic computation, symbols and groups of symbols are manipulated as if they were words and sentences. Of course the symbols and symbol groups do not have to reflect a natural language, they might reflect other languages as well, for example those of mathematics or music. For historical reasons, symbolic expressions came to be divided into two datatypes that reflect the basic distinction between words and groups of words. An *atom* is a basic, elementary thing like a symbol or a number. A *list* is group of things. A list can hold atoms and other lists. Lists are the central hallmark of the Lisp language. In fact, the name “Lisp” is an abbreviation for List processing. The following graph depicts the most common types of symbolic expressions, or *s-expressions* that will be encountered in this book.

```
              symbolic expression
                 /          \
              atom          list
             /    \
        number     symbol
       /  |   \
 integer float ratio
```

The symbolic expressions listed in the graph are actually just a tiny fraction of the total number of datatypes represented in Lisp. Lisp also provides the means to define new types of data and add them into the language.However, the graph contains the most basic and important set of datatypes that the Lisp composer will work with on a daily basis. The following sections of this chapter introduce these datatypes, provide examples of their notation and discuss their possible uses in music programming.

## Integers

An integer is an exact counting number like **`1`**, **`0`** or **`-99`**. Integer digits can be optionally preceded by a **`+`** or **`-`** sign. The special prefix **`#x`** allows an integer to be notated in base 16 (hexadecimal), **`#o`** is the prefix for octal notation and **`#b`** for binary. For all practical purposes there is no limit on the size of an integer in Lisp.

Integers are a basic feature of almost all programming languages. In addition to their role in mathematical expressions, many programs use integers for counting the number of times some action is performed. In terms of purely musical information, integers are a natural choice to represent pitch classes and interval distances expressed as half steps. Integers are also ideal for representing MIDI information such as *key numbers*. A key number is an integer between 0 and 127 that represents the position of a key on a "virtual" musical keyboard. The first key on the keyboard is key number 0; Middle C on the keyboard is key number 60.

Example 3-1. Samples of integer notation.

```
44100

-1

102003012303023024402323133293823 81

60

#xFFFF

#b-11110000
```

## Real Numbers

A real, or floating point number, is a fractional number like 1.25 or -.45. Very large or small real values can be written using exponential notation. In exponential notation the real number is followed by a suffix e+*n* or e-*n* where *n* is the appropriate power of ten. For example the large real number 1230000000000.0 can be written more concisely as 1.23e+12 in exponential notation.

Real numbers, like integers, are ubiquitous in programming. But unlike integers, there are a few issues to keep in mind when using real numbers. First, a real value is an inexact representation of a quantity. For example, 0.333 is only an approximation of the quantity 1/3. The value 0.333333 is a closer approximation to 1/3 than 0.333 but it is still not exact. A real approximation can only go to a certain *precision* because there is a limit to the size of floating point numbers in a computer. The maximum precision and size of a real number depends on each Lisp implementation. Secondly, when an inexact number is used in a calculation a slight error is introduced into the result. If this result accumulates over many repetitions of a computation then what was originally an insignificant difference may compound itself into a serious error in the final result. And finally, mixed arithmetic — mathematical expressions that contain both real and rational numbers (integers and ratios) — will always produce a real result. Multiplying 10 by 1.0 produces 10.0 not 10. This conversion into the inexact representation happens automatically and so it may be the source of confusion for the novice programmer.

Despite these caveats real numbers are a very effective way to represent measurements, and many types of musical quantities that only need to be "good enough". These include such things as frequency in Hertz and time values in seconds. Values like Hertz and seconds may only need to be "good enough" because humans can only discriminate their precision up to a certain point. Real numbers between 0.0 and 1.0 can also be used in composition to represent "normalized" values. A normalized value is like a percentage except that it may be directly scaled by another quantity to produce the desired value.

Example 3-2. Samples of real number notation.

```
440.0

4.3E+2
```

```
-.25

33.33333333333333

6.5E-10
```

## Ratios

A ratio is a fraction of two integers, for example 1/2 or -3/4. The numerator and denominator are separated by a `/` with no spaces. Lisp automatically reduces ratios to their simplest canonical form, for example 2/4 becomes 1/2 and 6/3 becomes 2.

Ratios are exact fractional quantities. They are useful in music for expressing exact proportions like those found in mensural notation, the harmonic series or even the Fibbonaci series. For example, if 1 represents a whole note then the ratio 2/3 is the proportion of 2 triplet half notes, 1/5 is 1 quintuplet quarter note, 7/8 is 7 eighth notes and so on. Ratios can also be used to represent intervals in the harmonic series and to produce *justly tuned* values by scaling (multiplying) a frequency *f* in Hertz . For example, 3/2*f* is the frequency a just fifth up from *f*, 8/9*f* is a just major second down, and so on. One potential problem with ratios is that sometimes they can be difficult to interpret. For example, how much is 160929076/151896821*f*?

Example 3-3. Samples of ratio notation.

```
13/21

5/3

-3/4
```

Lisp Note:

> In Common Lisp ratios are an exact representation. Guile converts ratio notation to inexact (real) numbers.

## Symbols

A symbol is a sequence of alphanumeric characters, like **`music`** or *5th-symphony*. Hyphenated symbols are very common in Lisp. Symbols can be thought of as words. Most combinations of letters, digits and hyphens can be included in a symbol's name. However, there are a few characters that have a special significance in Lisp notation and should not be used in symbol names. These characters are:

```
` # ( ) \ : ; " ' ,
```

Lisp can be either case sensitive or case insensitive. Case sensitive means that a lower case letter is not recognized to be the same as its upper case equivalent. In a case sensitive Lisp the three symbols **`machaut`**, **`Machaut`** and **`MACHAUT`** would all be different symbols. In a case insensitive Lisp the three notations would refer to the exact same symbol. For simplicity's sake, and because standard Lisp programming style uses lower case letters exclusively, this book assumes a case sensitive lisp and symbols are always notated in lower case letters.

Symbols appear everywhere in Lisp. In addition to their role as words and tokens they have a special significance as names for Lisp programs and variables. A *keyword* is a special type of symbol that starts with a colon, for example **`:retrograde`**. The colon is not actually part of the symbol's name, it is a special character that defines the symbol to be a keyword. Keywords have a special role in programming that will be discussed in the next few chapters. For now just remember that keywords can be used as tokens just like regular symbols can be.

Example 3-4. Samples of symbol notation.

```
c-sharp-minor

+

*tempo*

sfz

:sounds-good
```

## Lists

How data are organized is often just as important as what the data represents! The idea of grouping things together to form larger units of structure permeates the activity of music composition and it is also a fundamental property of Lisp data representation. All but the simplest programs in this book exhibit data grouping that forms higher level units of information out of the constituent elements. For example, a twelve-tone composition project might use integers to represent the twelve pitch classes and then group pitch classes together to form “rows”. The twelve-tone program would then operate on each row as a single unit of information. Note that a grouping like a twelve-tone row is homogeneous — all of its elements are of the same type and represent the same type of information. But this need not be the case — grouping can reflect an association between complex and heterogeneous information as well. For example, a project that involves Forte sets might represent each set as a group of three elements: the name of the set, the prime form of the set and the interval vector of the set. Each of these values means something quite different in the representation and would likely be implemented using different Lisp datatypes. To understand more about how data can be grouped into bundles to form “higher level” representations requires some knowledge about how Lisp itself represents and organizes data.

A list is a container. The contents of a list are called its elements. Any kind of data, including lists, can be placed inside a list. Lists are an extremely versatile tool; their recursive nature means that there is practically no limit to the depth and complexity of information that they can represent. A few very simple formats for organizing list elements have proven to be especially effective for representing many different types of information. The most basic of these formats is called the property list. The property list format will be discussed in a later section of this chapter.

It is not possible to categorize all the ways in which lists can be used in music composition — sets, rows, notes, chords, measures, phrases, sections, movements, pieces, staves, sequences, matrices, envelopes — the list goes on and on. But before learning how musical information might be represented with lists it is necessary to understand how a list is notated.

### List Notation

Lists are beautiful. They are simple, powerful, flexible and extremely easy to work with. There are only four very simple rules to proper list notation:

1. The starting and ending points of a list are marked by parentheses: **()**. The left, or open, parentheses **(** marks the start of the list and the right, or close, parenthesis **)** marks its end.
2. Every open parenthesis must be matched, or balanced, by exactly one close parenthesis.
3. A close parenthesis matches the nearest unmatched open parenthesis to its left.
4. Spaces, tabs or lines separate elements in a list.

It is very important to realize that the ability to understand and implement Lisp music programs depends on the ability to read and write lists. So if you understand the preceding rules then you are ready to start programming!

Example 3-5. Samples of list notation.

```
()

(0 3 7 11 2 5 9 1 4 6 8 10)

(how many violists does it
     take to screw in a light bulb?)

(3-7 (0 2 5) (0 1 1 0 1 0))

(staff violin-1
       (clef treble)
       (signature c-minor)
       (meter 2 4)
       (measure (rest 8th)
                (note g4 8th ff)
                (note g4 8th)
                (note g4 8th))
       (measure (note e4 half fermata)))
```

The first list in the example has no elements. A list with no elements is called the empty list. In other words, the empty list is still a list — a list without elements is not the same as no list at all. The second list in the example has 12 elements, each element is an integer. This list is one possible way to represent the twelve tone row from Berg's Violin Concerto. The third list demonstrates how sentences can be represented by symbols grouped into lists. The fourth list demonstrates one possible representation of a Forte set. It includes the name, prime form and interval vector of [025]. The last list in the example shows one possible way to represent the opening violin part of a very famous symphony.

## Property lists

Lists are collections that associate or group information together. Although lists can reflect a multitude of different organizations there are a few list formats that have proven to be very effective for representing different types of information. The most important of these list formats is called the property list. A property list is a list whose elements are interpreted pair wise. The first element of each pair is called the property and the second element is called the value. Since a pair consists of two elements a property list will always have an even number of elements. The entire list can be interpreted as the description of the properties, or attributes of some entity. How might this be used to represent music? Consider a program that computes flute compositions. Each note for this composition might be defined to have the acoustic properties of duration, frequency, amplitude, and timbre. Given this set of properties, individual notes in a composition would be specified by computing the appropriate values for each property of each note. For example the program might start out the composition by computing the following note description:

Table 3-2. Property and value pairs.

| Property | Value |
|---|---|
| duration | 7/8 |
| frequency | 440 |
| amplitude | ff |
| timbre | breathy |

The information in this table is easily represented in the composition program as a property list:

Example 3-6. Property list notation.

```
(duration 7/8 frequency 440
              amplitude ff timbre breathy)
```

It is quite common for keywords (symbols that start with ":") to be used as the property names in lists such as the one shown above. There are several reasons for this. First, using keyword names gives a visual cue as to what is serving as name and what is value in a property list:

Example 3-7. Keyword names in property lists.

```
(:duration 7/8 :frequency 440
           :amplitude ff :timbre breathy)
```

Secondly, some functions we will encounter later in this book allow *keyword arguments* to be passed to the function. A keyword argument is an optional input value that is referred by its keyword name. The syntax for these arguments shares the same keyword and value pairing as seen in the list above.

## Function call lists

Passing inputs to a function and receiving its output is referred to as *calling* a function, or *applying a function to its arguments*. Function calls follow a very consistent format in Lisp: a function call is a list whose first element is the name of the function and the rest of the elements (if any) are *arguments* to the function. Arguments are the data passed to the function as input.

This type of notation — in which the function always appears before the data passed to it — is called *prefix notation*. Prefix notation makes it easy to determine when symbols are used as functions and when they are not: for a symbol to act as a function it must always be in the first position of a list. In Example 3-8, the symbols **`*`** **`list`** and **`expt`** are functions because each is in the first position of a list. The symbol **`pi`** is not used as a function because it is the not the first element of a list.

Example 3-8. Function call notation.

```
(* 2 pi)

(list 2 pi)

(* 440 (expt 2 1/12))
```

Note that function calls may be *nested*. In the third example the second argument to the **`*`** is itself a function call. Embedding function calls inside other function calls is very typical of Lisp programming style.

## Other Common Datatypes

This chapter has introduced the basic building blocks of symbolic expressions — numbers, symbols and lists — that are used by the composition programs in this book. There are two other datatypes that historically were not considered to be symbolic expressions but nevertheless appear in many of the programming examples in this book.

## Boolean

Both the integer and real datatypes represent an infinite number of values. The boolean datatype represents only two possible values: true and false. Computer programs use boolean values to express *conditional actions*, actions that are only performed if a certain test is either true or false. For example, a music composition program might start by first randomly generating interval sets until a set with a certain interval vector content has been generated. If that test is true then the program begins to compose using the set, otherwise (the test was false) the program keeps picking random interval sets until it finds one that it "likes". Logic tells us that if something is not false then it must be true. In other words, false is false and true can be expressed as "not false". Some computer programs in Lisp do not test for an explicit boolean true value, they simply check to see that values are not boolean false. This allows *predicate functions*, functions that test the truth or falsity of an input expression, to return values other than explicit boolean true and false values.

Example 3-9. Samples of boolean notation.

```
#f

#t
```

Lisp Note:
: Scheme defines **`#t`** as boolean true and **`#f`** as boolean false. Common Lisp defines the symbol **`t`** to mean true and the symbol **`nil`** to mean false. This book generally refers to these values as simply true or false.

## Text Strings

A string is a sequence of text characters delimited by double quotes at the start and end of the string, for example **`"Music"`** or **`"Das Lied von der Erde"`**. Strings are used primarily for printing messages to windows or printing text to files stored on the computer. They are also used to specify file names and file directories on the computer. Strings are simply blocks of text, they have no other significance to Lisp. The text within the string may contain any combination of characters, even the special characters listed in the section describing symbols. To include the **`"`** character as part of the text it must be preceded by the special *escape character* **`\`** which tells lisp that the quote following the escape character is not interpreted as the end of the string.

Do not confuse strings with symbols or lists. "Music" is word-like but the pair of double-quotes makes it a string not a symbol. "Music is an art" contains words separated by spaces but it is not a list. These differences can be seen in the following summary table:

Table 3-3. Strings, symbols and lists.

| | |
|---|---|
| symbol: | **`music`** |
| text string: | **`"music"`** |
| list of symbols: | **`(music is an art)`** |
| text string: | **`"music is an art"`** |
| text string: | **`"(music is an art)"`** |
| list of strings: | **`("music" "is" "an" "art")`** |

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# 4 *Etudes*, Op. 1: The Lisp Listener

*Etudes* Op. 1 introduces the Common Music application and the Lisp Listener. We will also look at remote evaluation from the text editor called Xemacs. Before the exercises in this chapter can be practiced the contents of the CD must be copied to the computer's hard drive. If the CD contents have not yet been copied to the computer's hard drive consult the Appendix "Installing the Software CD" before continuing.

## Op. 1, No. 1: Starting the Common Music Application

To start Common Music locate the directory that contains the contents copied from the CD. This directory will always be referred to as the “NM project directory”. Once the NM project directory has been located the Common Music application can be started by performing the instruction with the icon that matches your computer's operating system below. There are two ways to start Common Music. The first method is to launch the application by double-clicking the 'cm' icon (application or script file) or executing the 'cm' script from a command shell. The second method is to launch the application from inside the Xemacs text editor. This second method allows you to edit lisp files and evaluate Lisp expressions without having to switch between two applications. On OSX, Windows and Linux, running Common Music inside Xemacs is the most efficient and interactive way to work with the system. Of course, it will require that you learn something about the Xemacs editor at some point. But this Etude we will simply learn how to select expression by clicking on them with the mouse and evaluating them using the Lisp menu.

MacOS Note:
    Double-click the Common Music application icon located in the NM project directory.
OSX Linux Note:
    Open a Terminal window and execute the 'xcm' script in the bin/ subdirectory of the NM project directory. This example assumes NM project directory was restored under the author's home directory:

```
/User/hkt/ ~/NM/bin/xcm
```

Windows Note:
    Double-click the file 'xcm.bat' located in the NM bin project directory.

Once the application launches a Lisp Listener window will be visible on the screen (Figure 1). Note that the appearance of window components such as borders, title bars, scroll bars and cursors will be different on each operating system.

## Op. 1, No. 2: The Lisp Listener Window

When a Lisp application starts running it opens a special window called the Listener. The Listener is a “viewport” into Lisp, it enables the user to interact with Lisp through a special program called the *interpreter*. The Listener window has several components, a menu bar, the main interpreter pane and a small mini buffer at the bottom of the window.

### The main menu bar

The application menu bar provides a set of menus to facilitate working with Common Music and Lisp. The File menu contains commands that open, create, load and compile Lisp files and quit the application. The Edit menu contains the standard set of text processing commands such as Cut, Copy and Paste. The commands in the Tools menu relate to programming in the Lisp environment and will not be covered in this practicum. The In/Out menu contains the interpreter's interaction history and allows past input expressions to be edited and re-entered into the

Listener. The Buffers menu selects between various windows that are currently open in the editor.

### The interpreter

The main working pane in the Listener window is called the Interpreter. The interpreter allows the user to interact with the Lisp application by entering Lisp expressions and observing the results. Input expressions can be typed directly into the window or sent remotely from inside the editor window. The interpreter and editor are introduced in the next two sections of this chapter.

### The mini buffer

The small region at the bottom of the window is called the mini buffer. The mini buffer is used to display status information and also acts as a prompt window in certain situations. The mini buffer will not be discussed further in this chapter.

## Op. 1, No. 3: The Lisp Interpreter

The main pane in the Listener window is called the Lisp interpreter. The interpreter allows the user to interact directly with Lisp by typing input expressions into the window and then observing what Lisp returns as an output value on the next line. You might think of the interpreter as a fancy calculator that determines the value of any Lisp expression you type to it. Whenever the interpreter is waiting for the user to enter an expression it displays the prompt **`cm>`** and the cursor bar just after it marks the insertion point, the position in the window where the next input expression should be typed. The prompt **`cm>`** is an abbreviation for Common Music, the Lisp application that is running inside the Listener. In this book the user's input expressions are always printed in bold face to distinguish them from the output values that Lisp prints back to the window. An output value always appears on the line directly beneath the input expression that produced it.

Click on the Listener window to make sure it is active and then type the following expression at the insertion point in the interpreter. Then press the Return key to enter the expression into Lisp. The Return key is not displayed in the examples, it is assumed that the user presses Return at the end of every input expression.

Interaction 4-1. Evaluating numbers.

```
cm> 440
440
cm> 44100.0
44100.0
cm>
```

As the user works inside the interpreter the entire interaction history is visible in the window. Although it is possible to type text anywhere inside the Listener only the insertion point at the very end of the last line in the window is active for a new line of input. Try entering a few different types of numbers into the interpreter: Next try entering a few symbols into the interpreter:

Interaction 4-2. Evaluating symbols.

```
cm> pi
3.141592653589793
cm> *scale*
#<tuning "standard-chromatic-scale">
cm> stravinsky
Error: Unbound variable: stravinsky
```

```
> While executing: "Unknown"
Aborted
cm>
```

Entering the first two symbols caused each variable's value to be printed (variables are discussed in the next chapter). Entering the fourth symbol **`stravinsky`** triggered a *Lisp error*.

## Op. 1, No. 4: Lisp Errors

Whenever lisp encounters a problem it prints an error message so that the user can address the problem. Each Lisp implementation has its own method for handling errors. Some versions of Lisp (for example Guile) may immediately return the user back to the Listener prompt after the error message is printed. Other implementations may enter an *error break* to allows a programmer to "inspect" the Lisp environment in order to find the problem that triggered the error. Error breaks are indicated by a message being printed in the Listener window and the Lisp prompt changing to a special break prompt. Break prompts often include a number that shows the current *error level* the user is working in. The error level indicates how many errors must be resolved before Lisp is back in its normal working environment. Lisp errors should never cause the application to "crash" or the computer to freeze. An error is really just a kind of report that attempts to explain what was wrong with the expression that caused the error. Note that while Lisp may be able to determine exactly what went wrong with an expression it cannot determine why something went wrong. This is because Lisp cannot know all the conditions that led to the problem. For example, when an *unbound variable* occurs in the Listener it may be due to the user misspelling the name of an actual variable or it may be the correct name of a variable the user forgot to define. Because Lisp cannot know the intentions of a user, error messages may seem cryptic at times, or may not give helpful information for correcting the problem. The novice Lisp composer should prepare for this eventuality by developing a consistent strategy for determining what caused the error and where that code is located in a program. This is called *debugging* a problem. Here are a few simple strategies to follow:

- Read the error message carefully and completely.
- Attempt to determine exactly what symbols or expressions are involved in the error.
- Locate these expression in the code you were just using.
- Study the code and its surroundings to see if you can identify the problem.
- If you cannot identity the problem, reduce the expression to smaller pieces until you have isolated the exact source of the problem.

There is practically no limit to the type and number of errors a human can make! The good news is that most problems are easy to resolve because they are simple mistakes like mistyping a name. The novice Lisp composer should pay particular attention symbol spelling and proper list formatting. Here is a list of the most common errors that the novice Lisp composer may make:

- Misspelled symbol
- Improperly formatted list
- Improper function call syntax
- Function call list whose first element is not a function
- Symbol or list not quoted

These issues will be discussed in more detail in Chapter 5.

## Op. 1, No. 5: The Lisp Editor

A composer spends most of the time in Lisp developing music programs. This activity is best done using a text editor that “understands” Lisp syntax. In this next exercise we will edit a sample Lisp source file and evaluate expressions in the file from inside the editor.

Choose the Open command from the File menu. Use Up and down arrows to navigate through the menu, press enter to select the file or enter a selected directory. When the dialog window pops up locate the file "opus-1.lisp" in the examples subdirectory of the NM project directory. Select the file. An editor window will open and display the contents of the file. Read the contents of the file and perform the suggested actions. Here is the contents of the example file.

Example 4-1. A Lisp file

```
;;; This is a sample lisp source file and you are reading
;;; a Lisp comment.  A comment begins with ; and continues
;;; to the end of the line. Lisp ignores comments when it
;;; process source code.

;;; To evaluate a lisp expression from inside the editor,
;;; first select the expression by double-clicking it and then
;;; use the Evel Expression command from the Lisp menu
;;; in the Editor's menu bar. Try evaluating the following
;;; expressions in the manner just described. After each
;;; expression is evaluated you will see its result printed
;;; in the Listener window.

2
pi
1/4

;;; To evaluate a list first double-click on one of the outer
;;; parentheses to select the whole expression. Then use Eval
;;; Selection to send the list to the Listener.
;;; You can also evaluate expressions by placing the cursor
;;; on the start or just after the expression and then
;;; typing the command: C-x C-e (control-x control-e).

(* 2 pi )

(list 2 pi)

(sin (* 2 pi ))

(define albans-row '(0 3 7 11 2 5 9 1 4 6 8 10))

(define (row->matrix row)
  ;; this function is discussed in Chapter 8
  (loop for i in row
        collect
        (loop for j in row
              collect (mod (+ (- i) j) 12))))

(pprint (row->matrix albans-row))
```

## Op. 1, No. 6: Improvisation

Try typing a few expressions of your own into the file and then evaluating them. Once you have finished experimenting with the sample code you can quit Common Music by selection "Quit" from the File menu in the

Listener window.

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file cmapp.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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H. Taube
06 May 2003

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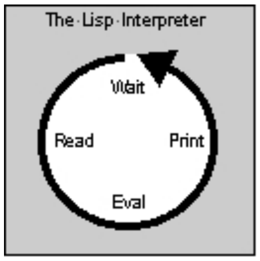
# 5 Lisp Evaluation

Understanding the process of Lisp evaluation is the key to understanding how Lisp programs operate and why Lisp programs are notated in the manner that they are. Lisp notation is simple to learn if the rules that govern evaluation are understood first. In this chapter we examine the process through which Lisp expressions are *interpreted*, or evaluated, to produce results. Since expressions are the building blocks of a Lisp program, in a very real sense the process of evaluation explains how Lisp operates as a programming language.

## The Interpreter

In the last chapter we learned how to type input expressions into the Listener window and to observe the results that Lisp printed back in the window. Lisp implements this interaction with a program called the interpreter, or *read-eval-print loop* (Figure 1).

Figure 5-1. The Lisp interpreter continuously processes input in its read-eval-print loop.



The terms *read*, *eval* and *print* refer the specific steps that the interpreter performs each time the user enters a Lisp expression into the Listener window. At the beginning of each cycle through the loop the interpreter waits for the user to input an expression. Once a user enters an expression — either by typing it and pressing the Return key or by remotely evaluating a block of text in an Xemacs buffer — the interpreter *reads* the input text, then *evaluates* the expression for content, and finally *prints* the results back to the window. To read and expression means that the input text is *parsed*, or assembled, into a Lisp expression. If a valid Lisp object cannot be parsed from the input then the interpreter signals an error. The eval step is the essence of Lisp. Evaluation means to determine a result, or *value* of a Lisp expression. Unlike the read and print step the Eval step cannot be seen in the Listener window which may make this process mysterious to the novice Lisp composer. The next section of this chapter explains the rules governing Eval's behavior and discusses Eval's effect on the datatypes that were presented in Chapter 2. The print step displays the results of evaluation in the Listener window. Printing is the reverse of reading, it involves translating the result expression back into a text string that is then displayed in the window.

## Evaluation

The purpose of Lisp is to determine the value of Lisp expressions. Evaluation is accomplished by applying "interpretation rules" to an expression. The rules that govern evaluation depend on the datatype of the expression. The easiest way to learn about evaluation is to observe its effects on the basic datatypes presented in Chapter 2.

### The Evaluation of Numbers, Booleans and Strings

Numbers, booleans and strings are *self evaluating* expressions. A self evaluating expression does not change as a result of evaluation. Common sense tells us that numbers and booleans must be self evaluating because, if they were not, the laws of mathematics and logic would make no sense — zero, one, true and false must remain ever so! Strings are self evaluating because Lisp defines strings to be blocks of uninterpreted text characters. The text inside the string has no significance whatsoever, the string is simply data.

Although the value of self evaluating expression remains unchanged by the evaluation process the result that the print step displays may appear different than the input expression. This difference does not mean that the value has changed, only that the print step has notated the value differently than the way the user entered it.

Interaction 5-1. Some self-evaluating expressions.

```
cm> 44100
44100
cm> #xFFFF
65535
cm> .500
0.5
cm> "Ondes Martenot"
"Ondes Martenot"
cm>
```

We can summarize our observations about self-evaluating expressions in a simple rule to remember:

Lisp Evaluation Rule 1:
    A number, boolean or string evaluates to itself.

## Symbol Evaluation and Variables

In Chapter Two we learned that symbols can be used as words and grouped into lists to form symbolic expressions. Here are two symbolic expressions that use the symbol **pi** differently:

```
        (pi is a greek letter)
        (* 2 pi)
```

In the first expression **pi** is used as the name of a Greek letter. But the second expression represents an *action*: "multiply 2 and pi together." In this case the symbol pi is clearly standing for a particular numerical value.

A symbol such as pi that names a value is called a *variable*. A symbol and a variable are not exactly the same thing. A symbol names a variable. A symbol can be created just by typing it but a variable can only be created by *defining* it using a special mechanism provided by Lisp. Although Lisp symbols appear to be just words they are actually complex objects that possess properties. One of the properties of a symbol is it's *print name*, the text that you see printed on the screen. A symbol also has a *value cell*. The value cell allows a value to be associated with a symbol. This association is called a *variable binding*. If a symbol is not bound before it is evaluated then Lisp will signal an "unbound variable" error. One of the interesting features of Lisp that distinguishes it from many other programming languages is the fact that a symbol's value cell can hold *any* Lisp value. Many languages constrain variables to hold only one particular type of data. But in Lisp it is the data itself that is typed, or classified, rather than the variable.

Interaction 5-2. Symbols are evaluated as variables.

```
cm> pi
3.141592653589793
cm> messiaen
> Error: Unbound variable: messiaen Aborted
cm>
```

In Interaction 5-2 the result of evaluating pi produced a value because the symbol **`pi`** was defined to represent that value. The evaluation of the symbol Messiaen signaled an error because it was not defined to have a value before it was evaluated. We summarize symbols as variables in a second rule of evaluation:

Lisp Evaluation Rule 2:
A symbol evaluates to its variable value.

Lisp signals an"unbound variable error" if the variable does not have a value.

## Evaluation of Function Call Lists

A *function call* is the application of a program to input data. Function calls are notated as lists: the first element of the list is the function and the rest of the elements are called the *arguments* to the function. In the expression:

```
(* 2 pi)
```

the symbol **`*`** is the function and **`2 pi`** are its two arguments. The evaluation of a function call requires that each argument to the function also be evaluated before the function is actually called. This process can be demonstrated using the example above. Before the multiplication can occur the two arguments 2 and pi are first evaluated in left to right order to produce the values that are to be multiplied. The first argument 2 is a number and therefore evaluates according to our Rule 1 to produce the value 2. The second argument pi is a symbol and so it is evaluated as a variable, according to Rule 2. The application of the second rule results in the value 3.141592653589793. Since neither argument produced an error, their values are passed to the function **`*`**, multiplied together and the result 6.283185307179586 is returned as the value of the function call.

Since each argument to a function is evaluated before the function is called, arguments may include variables and other function calls. It is very common to see *nested* function calls in lisp programming. A nested function call is an argument to a function that is itself a function call. In last input expression in Interaction 5-3 the function call **`(* 2 pi 1/4)`** is nested as the argument to the **`sin`** function.

Interaction 5-3. Function call notation.

```
cm> (* 2 pi)
6.283185307179586
cm> (list 2 pi)
(2 3.141592653589793)
cm> (sin (* 2 pi 1/4))
1.0
cm>
```

Each of the input lists in are *function calls*. The first element of a function call list is a function and the rest of the elements are arguments to the function. In the expressions above the symbols **`*`**, **`list`** and **`sin`** are used functions because they are symbols in the first position of an unquoted list. The symbol **`pi`** is not a function because it is not the first element of a list. We summarize function call lists in a third evaluation rule:

Lisp Evaluation Rule 3:
A Lisp list is evaluated as a function call.

The first element in the list must be a function and the rest of the elements are arguments to the function. Lisp signals an error if the first element is not a function of if any argument cannot be evaluated. Arguments in a function call are evaluated in left to right order to produce inputs to the function. The inputs are passed to the function and the result returned by the function becomes the value of the function call expression.

## Blocking Evaluation

The previous two sections have described how the evaluation process treats symbols and lists. This brings up the obvious question: how can a symbol or list ever be used as *data* in the evaluation process? That is, how can symbol be used as a token or a list as a data structure if the evaluator treats them as programming variables and function calls? For example, how can **`pi`** ever be used simply word as a word? and a list as a list is always interpreted as a function call then how can a list ever be data, for example, a 12-tone row?

The answer to both these questions is that the Eval step provides a simple way to selectively “turn off” the evaluation of any input expression that is meant to be interpreted as just data.

### The Lisp Quote

The single quote character: `'` is special character that “defeats” evaluation by forcing Lisp to accept the expression exactly as the user typed it. The Eval step evaluates a quoted expression by stripping off the quote and returning (now unquoted) expression as the result. The quote character can be applied to any Lisp expression. In the case of numbers, booleans and strings the quote will have no effect since these types of data are self evaluating anyway. A quote in front of a list of numbers allows that list to represent a grouping of data, for example, a twelve-tone row. A quote before a function call list will cause that list to become data rather than a function call.

Interaction 5-4. The effect of quote on evaluated expressions.

```
cm> '44100
44100
cm> '"Ondes Martenot"
"Ondes Martenot"
cm> pi
3.141592653589793
cm> 'pi
pi
cm> ''pi
'pi
cm> '(0 2 5)
(0 2 5)
cm> (row->matrix '(0 2 5))
((0 2 5) (10 12 15) (7 9 12))
cm> (* 2 pi)
6.283185307179586
cm> '(* pi)
(* pi)
cm>
```

Lisp Evaluation Rule 4:

A quoted expression evaluates to the expression without the quote.

### To quote or not to quote...

The novice Lisp composer can easily become confused about when and when not to use quote in the Lisp Listener. Although no set of rules can cover all possible situations there are a few simple guidelines that should help make this determination less confusing:

- Numbers never need to be quoted.

- Booleans (true and false) never need to be quoted.
- Strings never need to be quoted.
- Symbols need to be quoted unless they have been defined as variables or they are *keywords* (symbols starting with the colon character)
- Lists need to be quoted unless they are function calls.

## The Lisp backquote

The Lisp *backquote* character: ` is related to the quote character. Rather than block evaluation altogether, the backquote performs *selective* evaluation, and is used to notate lists in which some elements are evaluated and others not. A backquoted list acts as a template, in which some elements are act as unevaluated constants while others elements are evaluated and *replaced* by their evaluated value in the list:

Interaction 5-5. The effects of quote and backquote on lists.

```
cm> '(a b pi c)
(a b pi c)
cm> `(a b ,pi c)
(a b 3.141592653589793 c)
cm>
```

The first input expression in Interaction 5-5 is a quoted list and Lisp returns the list as the evaluated result. The second input expression is a backquoted list that selectively evaluates the variable pi and replaces it with its evaluated result in the list. A backquoted list acts just like a quoted list except that expressions preceded by a comma "," are explicitly evaluated. The backquoted list is particularly useful for notating lists that will contain many unevaluated tokens and a few expressions that must be evaluated. Since each unevaluated expression would otherwise have to be quoted in an evaluated list, the backquote can substantially reduce the size and complexity of a notated expression by eliminating the need for explicit quotes. The following two expressions are equivalent:

Interaction 5-6. Equivalent expressions using list and backquote

```
cm> (list 'a 'b pi 'c)
(a b 3.141592653589793 c)
cm> `(a b ,pi c)
(a b 3.141592653589793 c)
cm>
```

# Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file eval.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 6 The Art of Lisp Programming

The process of preparing programs for a digital computer is especially attractive, not only because it can be economically and scientifically rewarding, but also because it can be an aesthetic experience much like composing poetry or music.

— Donald E. Knuth

Programming in Lisp means to *implement*, or write, a series of Lisp expressions that — when evaluated — produce an intended result. In the case of this book, the intended result will more often than not be of a musical nature. Once the goal of a programming task is understood, implementing its code really consists of only three things: defining variables to hold data, defining functions that operate on data, and calling functions to produce the intended results. The clear, consistent, and functional nature of Lisp makes this process easy to learn. The following little music program will likely be understood by the reader even before the specific content can be explained. (Looping is discussed in Chapter 7.)

Example 6-1. A small Lisp program.

```
(define middle-c 60)

(define albans-row '(0 3 7 11 2 5 9 1 4 6 8 10))

(define (transpose-row row offset)
  (loop for pc in row collect (+ pc offset)))
```

After studying the example, start the Common Music application and evaluate each expressions in the Listener to see what **`transpose-row`** returns. Try calling the program several times

Interaction 6-1. Using the program

```
cm> (transpose-row albans-row middle-c)
(60 63 67 71 62 65 69 61 64 66 68 70)
cm> (transpose-row '(0 2 5) 90)
(90 92 95)
cm> (transpose-row albans-row (random 80))
(17 20 24 28 19 22 26 18 21 23 25 27)
cm>
```

## Programs are Functions

Because all actions in Lisp are accomplished by calling functions, learning how to write music in Lisp really amounts to learning how to call functions and knowing which functions to call in a given situation. A large part of this learning process is simply becoming familiar with the most frequently used functions encountered in music programming. To become familiar with a function means to learn a function's name, to understand what the function does, and to know how to call the function with appropriate input values. This chapter introduces a core set of Lisp functions that the composition programs presented in the rest of this book depend on. These functions include the basic arithmetic and boolean operators, list functions, and the special function **`define`** that allows new variables and functions to be added to Lisp. The functions that are presented here are only only a small portion of the total number functions available to you when in Lisp, and you should consult the Language specification for more information as you become familiar with each operator. The language specifications for Scheme and Common Lisp are indispensable as learning tools for both beginning and advanced Lisp users. One quick way to gain familiarity with Lisp is to look up a definition of interest in the language specification and to read through all of the entries in that section of the book. The next chapter presents some small, simple music programming examples that use the core functions. The reader is also strongly encouraged to browse through one or more of the Lisp reference books listed in the appendix.

## Function Names and Function Types

The name of a function will usually give a good indication of what its purpose is. For example, the function **`+`** calculates the sum of its input values. Lisp programmers tend to follow a set of naming conventions that help provide information about the role a function plays in programming. It will be helpful to keep some of these conventions in mind as the composer is introduced to new functions in the coming chapters.

A *predicate* is a function that returns true or false based on a test it applies to its input. Predicates can usually be identified by the suffix **`?`** appended to their names. For example, the predicate **`list?`** returns true if its argument is a list otherwise it returns false. The function **`odd?`** is a numerical predicate that returns true if its input is an odd number.

A *getter* is a function that returns a value from a Lisp object. For example, the getter **`first`** returns the first element in a list. Getters may have the suffix **`-ref`** appended to their names ("ref" is shorthand for "reference"). For example, the function **`list-ref`** returns an element from a specified location in a list.

A *setter* is a function that sets a value in a Lisp object. For example, the setter **`list-set!`** puts a value at a specified location in a list. Setters almost always have the suffix **`!`** appended to their name. The **`!`** informs the user that the function literally alters the object that it sets.

A *constructor* is a function that creates a Lisp object. For example, the constructor **`list`** creates a Lisp list out of the values specified to the function. Constructors sometimes have the prefix **`make-`** added to their name. For example, the constructor **`make-list`** creates a list of a specified length.

A *converter* is a function that converts one type of Lisp object into another type. For example, the converter **`symbol->string`** returns a Lisp string containing the name of a specified symbol. Converter names usually contain the "arrow" **`->`** between their input and output types.

## Function Call Syntax

Every Lisp function has a particular convention for accepting input values. Most functions expect a fixed number of input values. Some functions support what are called *optional arguments* — arguments that may be specified or not in a function call. A few common functions like **`*`**, **`+`** and **`list`** allow any number of arguments to be specified in a function call, including none.

# The Core Functions

The following sections in this chapter present an overview of the most basic functions encountered in the programs that appear in this book. This core set includes arithmetic operators, list functions and the special function **`define`** that adds variables and functions to Lisp. Each function entry documented below includes the name of the function, its function call syntax, and a brief description of what the function does. The function call syntax is depicted using a generalized description of what an actual function call would look like, including the type and number of arguments the function accepts. Arguments to the function are shown as italicized "meta-variables", like *x* that stand for actual argument data specified to the function. The special marker ... means that any number of additional input arguments may be specified from that point on in the function call. An input variable enclosed in square brackets, like [*arg*], means that the enclosed term or terms are *optional*. An optional argument is an argument that can appear or not in the function call. If it does not appear the optional argument has a *default value* that the function uses. Unless otherwise specified, the default value of an optional argument is boolean false.

## Numerical Functions

Lisp provides many numerical functions, only the most common are listed here. Consult the Scheme or Common Lisp references for the complete listing.

### Numerical Predicates

[Function]
    (**number?** *x*)
[Function]
    (**even?** *n*)
[Function]
    (**odd?** *n*)

The numerical predicates return boolean true or false based on a test of their arguments. **number?** returns true if *x* is a number, otherwise false. **even?** is true if *n* is an even number and **odd?** is true if *n* is odd.

Interaction 6-2. Examples of numerical predicates.

```
cm> (number? 1)
#t
cm> (number? '(1))
#f
cm> (even? 1.0)
#f
cm> (odd? -3)
#t
cm>
```

### Arithmetic Relations

[Function]
    (**=** *n* ...)
[Function]
    (**<** *n* ...)
[Function]
    (**>** *n* ...)
[Function]
    (**<=** *n* ...)
[Function]
    (**>=** *n* ...)

The arithmetic relations return boolean true if the relation is true for all arguments, otherwise false. Inequalities of more than two arguments test the order of the values.

Interaction 6-3. Examples of arithmetic relations.

```
cm> (= 1 1.0)
#t
```

```
cm> (< 1 1)
#f
cm> (> 3 2 1)
#t
cm> (>= -1 0 1 1 100)
#f
cm> (<= -1 0 100 .3)
#t
cm>
```

## Arithmetic Operators

[Function]
 (**+** *...*)
[Function]
 (***** *...*)
[Function]
 (**–** *n ...*)
[Function]
 (**/** *n ...*)

Return the sum, product, difference and quotient of their arguments, respectively. **+** and ***** return the operator identity (zero and one, respectively) if no arguments are specified. **–** returns the negative inverse of one argument or subtraction of two or more arguments. **/** returns the reciprocal of one argument or quotient of two or more arguments.

Interaction 6-4. Example arithmetic operators.

```
cm> (+ 1 2 3 4)
10
cm> (* 10 10)
100
cm> (- 1)
-1
cm> (- 1 .5)
0.5
cm> (/ 1)
1
cm> (/ 1 .25)
4.0
cm> (/ 30 5 -2)
-3
cm> (+)
0
cm> (*)
1
cm>
```

## Other Numerical Functions

[Function]
 (**min** *n1 n2 ...*)
[Function]
 (**max** *n1 n2 ...*)
[Function]
 (**mod** *n divisor*)

[Function]
    (**expt** *n power*)
[Function]
    (**log** *n*)
[Function]
    (**sin** *n*)
[Function]
    (**cos** *n*)
[Function]
    (**random** *n*)

**min** and **max** return the smallest and largest of their arguments, respectively. **mod** returns *n* modulus the *divisor*. **expt**returns *n* raised to the exponent *power* and **log** returns the exponent of *n* base *e*. **sin** and **cos** are the trigonometric functions sine and cosine, respectively. Both **sin** and **cos** take a single argument *n* in radians. The **random** function returns a uniform random number between zero less than *n*. If *n* is an integer then **random** returns an integer, if *n* is a real then a real value is returned. This means the **random** function can generate values in discrete and continuous distributions.

## List Functions

Lists are an essential feature of Lisp and are used everywhere in this book. The arithmetic and list functions will form the core of your programming vocabulary. Take the time necessary to work with functions in the interpreter until you are familiar with their behavior.

### List Predicates

[Function]
    (**list?** *x*)
[Function]
    (**null?** *x*)

The **list?** predicate returns true if *x* is a list and **null?** returns true if *x* is the empty list.

Interaction 6-5. The list predicates.

```
cm> (list? '())
#t
cm> (list? 1)
#f
cm> (list? '(1))
#t
cm> (null? '())
#t
cm> (null? '(1))
#f
cm>
```

### List Constructors

[Function]
    (**list** ...)
[Function]

(**`list*`** *x list*)

[Function]

(**`make-list`** *len*)

**`list`** creates and returns a list of its arguments. **`list*`** returns a list with *x* added to the front of *list*. **`make-list`** creates a list of length *len* with list elements initialized to false.

Interaction 6-6. List constructors.

```
cm> (list)
()
cm> (list 1 2 3)
(1 2 3)
cm> (list* 1 '(2 3))
(1 2 3)
cm>
```

## List Getters

[Function]

(**`first`** *list*)

[Function]

(**`second`** *list*)

[Function]

(**`third`** *list*)

[Function]

(**`fourth`** *list*)

[Function]

(**`rest`** *list*)

[Function]

(**`list-ref`** *list index*)

The functions **`first`** through **`fourth`** return the first, second, third and fourth elements in a list. The function **`rest`** returns all but the first element in *list*. The function **`list-ref`** returns the element at the *index* position in *list*. The first element in a list is at index zero.

Interaction 6-7. List accessors.

```
cm> (define my-notes '(a b c d e))

cm> (first my-notes)
a
cm> (rest my-notes)
(b c d e)
cm> (third my-notes)
c
cm> (list-ref my-notes 0)
a
cm> (list-ref my-notes (random 5))
d
cm>
```

## List Setters

[Function]
    (**`list-set!`** *list index x*)

Sets the element at position *index* in *list* to *x*.

Interaction 6-8. List setting

```
cm> (list-set! my-notes 0 'f)

cm> my-notes
(f b c d e)
cm>
```

Take care using setters as they "destructively" modify, or alter, what they set. We will not have occasion to use setters very often in this book.

## Other List Functions

[Function]
    (**`length`** *list*)
[Function]
    (**`reverse`** *list*)
[Function]
    (**`append`** *...*)
[Function]
    (**`member`** *x list*)
[Function]
    (**`position`** *x list*)

**`length`** returns the number of elements in *list*. **`reverse`** returns a copy of *list* with its elements in reverse order. The **`append`** function concatenates all arguments into a single list. Each argument must be a list. **`member`** returns true if *x* is an element of *list*, otherwise false. **`position`** returns the position of *x* in *list* or false if *x* is not in the list.

Interaction 6-9. List setting

```
cm> (define my-notes '(a b c d e))

cm> (length my-notes)
5
cm> (reverse my-notes)
(e d c b a)
cm> (append my-notes (reverse my-notes) my-notes)
(a b c d e e d c b a a b c d e)
cm> (member 'g my-notes)
#f
cm> (member 'e my-notes)
#t
cm> (position 'b my-notes)
1
cm> (position 'g my-notes)
#f
cm>
```

## Boolean Operators

The boolean operators return either true or false based on a test of their arguments. Remember that Lisp considers any value to be "true" unless the value is strictly boolean false. Scheme and Common Lisp use different values to designates boolean true and boolean false.

[Function]
(**and** *...*)
[Function]
(**or** *...*)
[Function]
(**not** *b*)
[Function]
(**if** *test then* [*else*])

The function **and** returns true if all its arguments are true, and **or** returns true if any of its arguments are true. **not** returns the logical negation of *b*. The **if** special function first tests the *test* expression. If it is true then the *then* expression is evaluated, otherwise an optional *else* expression is evaluated.

Interaction 6-10. Example of boolean operators.

```
cm> (and #t #t #t)
#t
cm> (and #f #t)
#f
cm> (and (= 1 1) (< 10 11))
#t
cm> (or #f #f #t #f)
#t
cm> (or #f #f)
#f
cm> (not #f)
#t
cm> (not (not #f))
#f
cm> (if #t 1 2)
1
cm> (if #f 1 2)
2
cm> (if (> pi 3) 'yes)
#f
cm> (if (< 1 3) 'winner 'loser)
winner
cm>
```

## Object Functions

Lisp *objects* are a way to represent user defined data and are the foundation for representing musical structure in Common Music. Objects and the functions that operate on them are introduced in Chapter 11.

## Input and Output

[Function]
(**print** *x*)

[Function]
    (**pprint** *x*)
[Function]
    (**describe** *x*)
[Function]
    (**load** *file*)

The **print** function prints *x* to the Lisp Listener window and returns *x* as its value while **pprint** "pretty prints" *x* by insuring that lists are displayed in a readable format. The **describe** function prints information about *x*. The **load** function loads *file* into Lisp and evaluates all of the expressions contained in the file.

## Sequencing

A number of special functions in Lisp *sequence*, or evaluate, a series of expressions one after the other. Several of these functions are discussed later in this chapter, we introduce only the most basic sequencing operator here.

[Function]
    (**begin** ...)

**begin** evaluates every argument in sequential order and returns the value of the last expression. Use **begin** to form a single expression out of a series of expressions.

Interaction 6-11. Sequential evaluation.

```
cm> (begin 1 2 3 'go)
go
cm> (if (< (random 3) 2)
      (begin (print 'winner)
             'payday)
      (begin (print 'loser)
             'go-to-jail))
loser
go-to-jail
cm>
```

**begin** was used in the **if** statement to link more than one expression in the *then* and *else* clauses. It appears as if the **if** expression returned two values, **loser** and **go-to-jail**. But actually, only the second value was returned. The first value (**loser**) was simply printed to the window by the **print** function.

# The define Special Function

Define is a *special function* that defines either variables or functions in Lisp.

## Defining Variables

[Function]
    (**define** *symbol value*)

Defines *symbol* as a variable bound to *value*.

Interaction 6-12. Defining variables.

```
cm> (define two-pi (* pi 2))

cm> two-pi
6.283185307179586
cm> (define lotsa-pi (* pi pi pi))

cm> lotsa-pi
31.006276680299816
cm> (define favorite-pie 'apple-crunch)

cm> favorite-pie
apple-crunch
cm>
```

If one studies these last interactions carefully it is possible to discover that **`define`** "breaks the rules" of Lisp evaluation. In the last chapter we learned that each argument in a function call is evaluated before the function is called. Indeed, the interactions above clearly show that the expressions (* 2 pi), (* pi pi pi) and 'apple-crunch were evaluated. But notice that the first arguments to **`define`** — the variable names — were *not* evaluated. If these symbols had been evaluated, all three expressions would have produced "unbound variable" errors according to Evaluation Rule 2. But no error occurred because **`define`** is a special function that does not evaluate its first argument. Instead, the first argument *names* the variable or function being defined.

Variables are defined for a number of different reasons. One reason is simply to give a name to a value. A name provides an "abstraction" from the raw data that indicates something about the role that the data plays in the program. Using the variable **`pi`** rather than the quantity 3.141592653589793 gives a much better indication of what role that value serves in a program. Of course, Lisp does not care what name is chosen for a variable — **`pi`** could just as well be called **`pie`** or **`hamburger`**. But taking time to select appropriate names is an important programming task because it helps to make code "self-documenting" and more understandable to other programmers who might read the code.

Variables can also be used to save the results of a calculation so that the same value does not need to be calculated more than one time. While in principle there is nothing wrong with recalculating the same value over and over again, every instruction that a computer executes takes some amount of time to perform. A program that recalculates the same value over and over again will take longer than it needs to without any benefit. In fact, if the calculations are complex (or if the computer is slow) the time wasted recalculating values can become a significant issue for the programmer.

A third reason to use a variable is, of course, to define a quantity whose value will change. Consider a program that generates twelve-tone music. If the program defines a global variable to hold the twelve tone row then new pieces can be generated from the same code simply by redefining the variable to hold a new row form. The **`define`** function can be used to define new global variables or to *redefine* an existing variable to a new value. When a variable is redefined its old value is simply "thrown away" and replaced by the new value.

Another way to specify a new value for an existing variable is to use the special *assignment* function **`set!`**:

[Function]
    (**`set!`** *symbol value*)

Sets the variable named *symbol* to *value*. The difference between **`define`** and **`set!`** is that **`set!`** simply assigns a variable a value, it does not actually declare the symbol to be a variable.

Interaction 6-13. Redefining and assigning variables.

```
cm> (define favorite-row '(0 7 8 3 4 11 10 5 6 1 2 9))

cm> favorite-row
(0 7 8 3 4 11 10 5 6 1 2 9)
cm> (define favorite-row (reverse favorite-row))

cm> favorite-row
(9 2 1 6 5 10 11 4 3 8 7 0)
cm> (set! favorite-row albans-row)

cm> favorite-row
(0 3 7 11 2 5 9 1 4 6 8 10)
cm>
```

## Defining Functions

[Function]

(**define** (*name* [*arg ...*]) *body ...*)

The definition of a function involves the specification of three pieces of information:

1. The name of the function.
2. The *input parameters* to the function, i.e. “windows” through which the function will receive input arguments.
3. The *body* of the function, the Lisp expressions that define the function's computational behavior.

A simple first example of a function definition will allow us to examine these points in greater detail:

Example 6-2. Convert beats per minute to time in seconds.

```
(define (bpm->seconds bpm)
  (/ 60.0 bpm))
```

This code in this example corresponds to the three parts of a function as follows:

1. Function name: **bpm->seconds**
2. Function parameters: **bpm**
3. Function body: **(/ 60.0 bpm)**

### Function Name

A function's name should reflect the programming task that the function implements. In our example, the name **bpm->seconds** was chosen because it tells the caller what the function does, namely, to convert a metronome value measured in beats per minute into time measured in seconds. Of course, other names are possible. Here are just a few alternatives that could also be used:

- **metronome-to-seconds**
- **beats-per-minute->seconds**
- **metronome->seconds**
- **metro->secs**
- **bpm2secs**

### Input Parameters

A function's input parameters are the “windows” through which the function receives input values from a function call. Each parameter can be thought of as a variable that is automatically bound to an argument value whenever the function is called. Statements inside the body of the function can then treat input values as defined variables. Once the function finishes executing the input parameters are automatically *unbound* such that their symbols no longer reference the values that were input to the function. Input parameters are called *lexically scoped variable*, which means that they only exist within the lexical (textual) confines of the function definition. For example, the function definition for **`metronome->seconds`** declares a single input parameter called *bpm*. This symbol will become a variable bound to the value specified as the argument to the function call. So in the following function call:

```
(metronome->seconds 60)
```

the *bpm* parameter (bpm means "beats per minute") to **`metronome->seconds`** is bound to the value of **`60`** while the function executes. After the function returns the symbol *bpm* is no longer bound as a variable.

### Function Body

The *body* of a function definition consists of one or more Lisp expressions that are evaluated in succession each time the function is called. The value of the last expression in the body determines the value of the function call. The body of the **`metronome->seconds`** definition consists of a single form:

```
(/ 60.0 bpm)
```

The effect of this form is to turn a metronome value like **`60`** into time measured in seconds:

Interaction 6-14. Examples of numerical predicates.

```
cm> (metronome->seconds 60)
1.0
cm> (metronome->seconds 120)
0.5
cm> (metronome->seconds 90)
0.6666666666666666
cm> (metronome->seconds 40)
1.5
cm>
```

## Local Variables

Sometimes a function requires *local variables* defined within the function to store values that the function uses in its calculation. Variables can be bound anywhere in a program using the **`let`** special function:

[Special Form]
  (**`let`** (*variable ...*) *body ...*)
[Special Form]
  (**`let*`** (*variable ...*) *body ...*)

Binds variables within lexical scope of the **`let`**. The first form after the name is a list of *variable bindings* Each binding is itself a list:

(*variable value*)

where *variable* is the name of the variable and *value* is its value. After the variable declarations comes one or more lisp expressions that constitute the body of the let. These expressions are evaluated in sequential order and the last expression becomes the value of the **let**. Example 6-3 shows several sample **let** statements:

Example 6-3. Three example let statements.

```
(let ()
  3)

(let ((a 10)
      (b 20))
  (* a b))

(let ((a (random 10)))
  (* a a))
```

The first **let** does not declare any variables and simply returns the value 3. The second **let** declares two variables *a* and *b* and binds them to the values 10 and 20, respectively. The third **let** binds a single variable *a* to a random number less than 10 and then returns the square of the value. Notice that in all cases a binding list is provided. If no variables are bound an empty bindings list is specified.

The body of a let behaves exactly the same as the body of a function definition: each expression inside the body is evaluated in sequential order and the last expression determines the value of the let expression. Here is an example of a let expression that establishes two variables, *a* and *b* then prints their values and returns their sum:

Interaction 6-15. A let definition with two variable bindings.

```
cm> (let ((a 10)
          (b (random 10)))
      (print a)
      (print b)
      (+ a b))
10
2
12
cm>
```

The first statement in the body of the **let** prints the value of *a*, the second prints the random value of *b* and the third expression adds the two variables together. All three expressions are evaluated in sequential order and the value of the last expression is returned as the value of the entire **let** statement. Note that it appears as if the let statement actually returned three values: 10 2 and 12. But only the last value was really returned by the **let**; the first two values were simply printed to the terminal by the **print** statement. The last value appears in the window because it was printed by the Listener's read-eval-print loop as the evaluated value of the **let** expression.

The only difference between **let** and **let*** is that **let** binds its variables in parallel while **let*** binds its variables in *sequential* order such that the binding of one variable can depend on the value of a preceding variable in the variable list. The following **let*** expression returns a list containing a randomly chosen MIDI key number and the key number one octave above it. **let*** is required because the second variable, *oct*, depends on the first variable, *key*, for its value:

Interaction 6-16. A let* definition with dependent variable definition.

```
cm> (let* ((key (random 116))
           (oct (+ key 12)))
      (list key oct))
(33 45)
cm>
```

In this example the *key* variable holds a randomly chosen MIDI key number. The upper bound for the random key choice is 116 because we do not want the upper octave value to ever exceed the maximum MIDI key number of 128. The *oct* variable is defined to be 12 semitones above the randomly chosen key value. Inside the body of the **`let*`** the **`list`** function creates a list containing the random key number and the octave above it. Since the list function call is the last form inside the let (it is actually the only form inside the let) the value returned by **`list`** becomes the value of the entire **`let`** expression.

Note that this last **`let`** expression must be evaluated a number of different times in order to see the effect of the random octave selection. Rather than input the same expression over and over again we will use this opportunity to define another function that simply *encapsulates* this expression so that it can be called by name. The function does not require any input arguments.

Example 6-4. Encapsulation in a function definition.

```
(define (random-octave)
  (let ((key (random 116)))
    (list key (+ key 12))))
```

Interaction 6-17. Calculating random octaves with the function.

```
cm> (random-octave)
(21 33)
cm> (random-octave)
(88 100)
cm> (random-octave)
(17 29)
cm>
```

Now that the calculation has been defined as a function the let expression has become a reusable and (potentially) useful compositional tool that can be employed in different musical programs simply by calling the function.

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file core.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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H. Taube
06 May 2003

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# 7 *Etudes*, Op. 2: Musical Math

Through and through the world is infested with quantity. To talk sense is to talk quantities, It is no use saying the nation is large- how large? It is no use saying that radium is scarce- how scarce? You cannot evade quantity. You may fly to poetry and music and quantity and number will face you in your rhythms and your octaves.

— Alfred North Whitehead

The Etudes in this chapter demonstrate how the core Lisp functions introduced in the previous chapter can be used to implement new functionality in Lisp. There are often many possible ways to implement a given function; the implementations used in this chapter were chosen more for their pedagogical clarity than anything else.

## Op. 2, No. 1: Rhythmic Time

In our first exercise we implement a function that converts metric rhythm values into time in seconds. We will use Lisp ratios to represent metric proportions. For example, a quarter-note will be notated as 1/4, a whole note as 1, three sixteenths is 3/16, and seven twenty-third notes would be 7/23. Since the basic metric proportions are relative to a whole note (at least for simple meters) to convert a metric value to seconds simply requires that the ratio be multiplied by 4.0, the length of a whole note in seconds. For example, 1/4*4.0 gives 1.0 seconds per quarter-note, and 3/16*4.0 give .75 seconds. Of course, every musician knows that to convert a metric proportion into seconds really requires a *tempo factor* in addition to a metric value. We will use the **metronome->seconds** function we defined in the last chapter for just this purpose. Here is the definition of **rhythm->seconds** again:

Example 7-1. Converting metric proportion to time in seconds.

```
(define (rhythm->seconds rhy tempo)
  (* rhy 4.0 (bpm->seconds tempo)))
```

When the tempo is 60 then **bpm->seconds** returns 1.0 and the value returned by **rhythm->seconds** will be unchanged. However, a tempo of 120 produces .5 as a tempo coefficient which scales the metric values returned by **rhythm->seconds** in half.

Interaction 7-1. Using the rhythm->seconds function.

```
cm> (rhythm->seconds 1/4 60)
1.0
cm> (rhythm->seconds 1/2 120)
1.0
cm> (rhythm->seconds 1/8 90)
0.3333333333333333
cm>
```

As an additional exercise, try defining a variation of this function that allows a beat value other than the whole note to serve as the basis for the calculation.

## Op. 2, No. 1: Transposing Hertz Frequency

In this next exercise we implement a function that transposes Hertz values by semitones, quarter tones, 13th tones, or any other equal division of the octave. In order to implement this function we must first understand something about the nature of frequency calculations. Every musician knows that an *octave* represents a doubling or halving of frequency. A sequence of frequencies spaced an octave apart can be calculated from a frequency *f* by simple multiplication: 1*f*, 2*f*, 4*f*, 8*f*, 16*f*, and so on. A process of halving the frequency *f*: 1*f*, 1/2*f*, 1/4*f*, 1/8*f*, 1/16*f*, will produce octaves downward. These doubling and halving sequences are *exponential progressions*, that is, each sequence can be expressed by powers of 2: $2^0$=1, $2^1$=2, $2^2$=4, $2^3$=8, $2^4$=16, and $2^0$=1, $2^{-1}$=1/2, $2^{-2}$=4, $2^{-3}$=1/8, $2^{-4}$=1/16. Since these are exponential progressions the Lisp function **`expt`** can be used to calculate octave sequences in either ascending or descending order, for example:

Interaction 7-2. Calculating octaves by powers of 2.

```
cm> (expt 2 -1)
.5
cm> (expt 2 -2)
.25
cm> (expt 2 0)
1
cm> (expt 2 1)
2
cm> (expt 2 2)
4
cm>
```

We can see in the previous example that as the exponents progress linearly from -2 to 2 the values returned by **`expt`** progress exponentially. since $2^0$ is the unison (1) and $2^1$ is the octave (2) then linearly increasing exponents that lie between 0 and 1 must determine frequencies within a single octave. Therefore the exact middle of an octave (tritone up) is $2^{1/2}$, 1 semitone up from *f* must be $f * 2^{1/12}$, 3 quarter-tones down from *f* is $f * 2^{-3/24}$, and so on:

Interaction 7-3. Exponentiation.

```
cm> (expt 2 0)
1
cm> (expt 2 1)
2
cm> (expt 2 1/2)
1.4142135623730951
cm> (expt 2 1/12)
1.0594630943592953
cm> (expt 2 -3/24)
0.9170040432046712
cm>
```

Given the **`expt`** function it is simple to define a function that scales any hertz value by any equal tempered amount:

Example 7-2. Scaling Hertz frequency values by equal tempered amounts.

```
(define scale-hz hz mul)
  (* hz (expt 2 mul)))
```

The **`scale-hz`** function simply multiplies an input Hertz value by an octave amount to transpose the value by an equal division of the octave:

Interaction 7-4. Calculating Hertz values with scale-hz.

```
cm> (scale-hz 440 1)
880
cm> (scale-hz 440 -2)
110
cm> (scale-hz 440 1/24)
452.8929841231365
cm>
```

## Op.2, No. 2: Converting MIDI Key Numbers to Hertz Frequency

In Example 7-3 we implement a function **`keynum->hertz`** to convert MIDI key numbers into equivalent Hertz frequencies in the equal tempered scale. Recall that MIDI key numbers represent positions in a 128 key *virtual* keyboard, where key number 0 is the lowest C (8.175 Hertz), 60 is Middle C (261.625), and 127 is the highest key (12543.853 Hz).

Example 7-3. Converting MIDI key numbers to Hertz frequency.

```
(define lowest-freq 8.175)

(define (keynum->hertz knum)
  (* lowest-freq (expt 2 (/ knum 12)
```

Since an octave contains 12 semitones, the octave exponent to **`expt`** is calculated by dividing the input key number by the number of keys in one octave.

Interaction 7-5. Converting MIDI key numbers to Hertz equivalents.

```
cm> (keynum->hertz 60)
261.6255653005986
cm> (keynum->hertz 69)
439.99999999999994
cm>
```

In the last expression of the preceding interaction the key number 69 returns 439.999, a value very close to, but not exactly, 440.0, the true frequency of tuning A. This small error is due to the fact that floating point numbers are inexact quantities and we are basing our calculations on a very low C. To implement a version of **`keynum->hertz`** in which Middle C is key number 60 but key number 69 returns exactly 440.0 requires that the Hertz calculation be based on a low A rather than a C, and that the key number difference between A and C (3 semitones) then be added to the input key number:

Example 7-4. A version of keynum->hertz based on A.

```
(define lowest-freq 6.875)

(define (keynum->hertz knum)
  (* lowest-freq (expt 2 (/ (+ knum 3) 12))))
```

Interaction 7-6. Converting MIDI key numbers to Hertz equivalents.

```
cm> (keynum->hertz 60)
261.6255653005986
cm> (keynum->hertz 69)
440.0
cm>
```

## Op.2, No. 3: Bounded Random Selection

Recall from the last chapter that the random function returns a value equal or greater than zero but less than the upper bounds specified to the function. In this next exercise we allow random selection between a lower and upper boundary specified by the user. If both boundaries are integers then **`pick-range`** returns a discrete (integer) value, otherwise a floating point number that lies between the low and high bounds is returned.

Example 7-5. Random selection between a low and high bounds.

```
(define (pick-range low high)
  (let ((rng (- high low)))
    (if (> rng 0)
      (+ low (random rng))
      0)))
```

The **`pick-range`** function defines both a lower and upper bounds for random selection. These values are converted to a range by subtracting *low* from *high*. If this range is greater than zero then **`pick-range`** selects a random value less than *rng* and adds it to *low* so that the value it returns lies between *low* and *high*.

Interaction 7-7. Examples of pick-range.

```
cm> (pick-range 60 100)
96
cm> (pick-range -1.0 1.0)
-0.697347603174574
cm> (pick-range 440 880)
789
cm>
```

## Op.2, No. 4: Random Selection from a List

This next function randomly selects an element from a list of values the user specifies to the function.

Example 7-6. Random selection from a list of values.

```
(define (pick-list lst)
  (let* ((len (length lst))
         (pos (random len)))
    (list-ref lst pos)))
```

**`pick-list`** selects an element by calculating a random position in the input list and returning the element at that position using the **`list-ref`** function. The **`let*`** special function allows the declaration of *pos* to reference the value of **`len`** in its binding. List positions always range from zero to one minus the length of the list. By specifying the length of the list as the exclusive upper bounds to **`random`** a valid position in the list is guaranteed to be chosen. The random element returned by **`list-ref`** becomes the value of the function call.

Interaction 7-8. Examples of pick-list.

```
cm> (define aeolian-mode
      '(57 59 60 62 64 65 67 69 71 72 74 3 77 79 81))

cm> (pick-list aeolian-mode)
62
cm> (pick-list aeolian-mode)
```

```
79
cm> (pick-list aeolian-mode)
71
cm> (pick-list '(1/16 1/8 1/8 1/4))
1/4
cm>
```

Note that in the definition of the rhythms list twice as many 1/8th note values are specified as 1/16th note or 1/4 notes. This is an easy way to implement *weighted random selection*, in which some outcomes have more probability of selection than others. The **`random`** function generates random numbers in a *uniform distribution* which means that there is an equal chance of any number in the range being selected. By adding one element multiple times there are more chances that one of the duplicate elements will be chosen.

## Op.2, No. 5: Chance and Probability

The function defined in this next exercise implements a predicate function based on a *probability factor* specified by the user. The probability factor is a real number between 0.0 and 1.0 inclusive, where a 0.0 probability means an outcome cannot happen and a probability of 1.0 means an outcome is certain to happen.

Example 7-7. Random selection from a list of values.

```
(define (chance? prob)
  (< (random 1.0) prob))
```

The **`chance?`** function returns true if a random choice is less than the probability *prob* specified to the function. If *prob* is 0.0 then there is a 0% probability that chance returns true because the random function never returns a value less than zero. If *prob* is 1.0 then there is a 100% probability of **`chance?`** returning true because **`(random 1.0)`** is guarantee to return a value less than 1.0.

Interaction 7-9. Examples of chance?

```
cm> (chance? 0)
#f
cm> (chance? 1)
#t
cm> (chance? .75)
#t
cm> (if (chance? .8) 'retrograde 'prime)
retrograde
cm> (if (chance? 0) 'retrograde 'prime)
prime
cm>
```

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file math.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 8 Iteration and the Loop Macro

Many programming tasks consist of one or more actions that are repeated over and over again. The process of repeating actions is called *iteration*, or *looping*. Lisp provides a powerful iteration facility called Loop to define iterative tasks. Loop is really more of a language for describing what an iteration does than it is a Lisp function. In order to introduce the expressive power of the Loop language we start by examining two very small music programs, both of which require iterative solutions.

## Iteration Examples

Consider a program that transposes twelve tone rows onto specific notes in the standard scale. Assume for the moment that notes are represented by MIDI key numbers 0-127 and that a twelve tone row is simply a list of pitch class integers 0-11. Data for this program could be defined using two variables, as shown in Example 8-1.

Example 8-1. Defining a transposition offset and a twelve tone row.

```
(define my-key 60)

(define my-row '(0 3 7 11 2 5 9 1 4 6 8 10))
```

The iterative task that our transposition program must implement can be expressed in a single sentence:

> "For each pitch class in *my-row*, add the pitch class to *my-key* and collect it into a new row."

This sentence is easily translated from English into Loop:

```
(loop for pc in my-row collect (+ pc my-key))
```

Interaction 8-1. Loop implementation of the transposition program.

```
cm> (loop for pc in my-row collect (+ pc my-key))
(60 63 67 71 62 65 69 61 64 66 68 70)
cm>
```

The **`loop`** function call in Example 8-1 can be seen to implement our row transposition. Within the **`loop`** expression, the *stepping variable* *pc* was automatically mapped over every pitch class in *my-row*. The function call **`(+ pc my-key)`** defines the operation that Loop applied to each pitch class in the row to effect the transposition. The **`collect`** command causes **`loop`** to gather the transposed values into a new list and to return that list as the value of the loop.

Our second example is a program that computes a list of Hertz frequencies using the harmonic series. The fundamental, starting harmonic number and ending harmonic number are defined in Example 8-2

Example 8-2. Defining the fundamental and bounds for a harmonic series.

```
(define fund 220)

(define harm1 1)

(define harm2 8)
```

The iterative task that this program implements can be expressed as follows:

"For each harmonic number from *harm1* to *harm2*, multiply *fund* by the harmonic number and collect it into a list."

This is implemented by the Loop expression:

```
(loop for h from harm1 to harm2 collect (* fund h))
```

Interaction 8-2. Loop implementation of the harmonic series iteration.

```
cm> (loop for h from harm1 to harm2 collect (* fund h))
(220 440 660 880 1100 1320 1540 1760)
cm>
```

Each of these examples consists of an operation that is repeated over a range of values. In the first example the operation (adding a transposition offset to a pitch class) is defined over elements in a list. In the second example the operation (multiplying a fundamental frequency by a harmonic number) is defined over consecutive integers between a lower and upper bound. The Loop language provided a consistent and simple interface to solving these different tasks. Note that in neither case did the programmer have to explicitly increment variables, or design predicate tests to determine when the iteration should stop, or construct the list that is returned from the loop. The loop language took care of these chores automatically from our description of the iteration.

## The Loop Facility

Loop is a *macro* that implements an English-like syntax for defining iterative tasks. The special syntax of **`loop`** makes it very easy for a programmer to implement different types of iterations, from simple “repeat” loops to a whole series of tasks performed in parallel. Note that — like the special function **`define`** — the loop syntax is so natural that we may not notice the fact that its syntax “breaks the rules” of normal Lisp evaluation. This can be shown by analyzing what should happen when the following loop expression is evaluated:

Interaction 8-3. A loop that sums numbers from zero to ten.

```
cm> (loop for i to 10 sum i)
55
cm>
```

If the regular evaluation rules of Lisp (Chapter 3) were applied to this function call, then Evaluation Rule 2 would signal an unbound variable error when the unquoted symbol **`for`** were encountered as the arguments to the function got evaluated (because **`for`** has not been defined as a variable.) No error occurs in Example 8-2 because **`loop`** is a special type of function called a macro (chapter 2) that is free to implement its own evaluation strategy. The **`loop`** macro implements *selective* evaluation, in which some of its arguments are evaluated and some are not. In particular, **`loop`** uses symbols like **`for`**, **`to`** and **`sum`** as *command words* and not as variables. Many introductory Lisp books and courses eschew **`loop`** precisely because it implements its own evaluation rules rather than following the normal model defined by Lisp. While this approach is admirably pure it means that the poor student must then learn out-dated and arcane methods to achieve effects that are more clearly expressed in the Loop language. This book expresses all iteration and mapping tasks using the Loop language. Loop is the exclusive choice for several important reasons:

- Loop's English-like syntax makes it very easy and natural to learn.
- Loop can express many different types of iteration.
- Loop is an excellent example of the ability to define different languages within Lisp.
- Loop's syntax is also used by the **`process`** macro introduced in Part 2 of this book.

- The process macro supports algorithmic generation of music. So if you can write a loop then you are able to compose algorithmic compositions.

## Loop Syntax

A **`loop`** expression consists of one or more *clauses*. A number of different types of clauses may appear in a single **`loop`** expression:

- *Initialization* clauses establish starting conditions for the iteration.
- *Stepping* clauses limit the iterative process and provide *stepping variables* whose values automatically change each time through the loop.
- *Action* clauses define task actions and allow values to be returned from the loop expression.
- *Conditional* clauses allow action clauses to depend on the truth or falsity of a test.
- *Finalization* clauses execute immediately after the iteration stops.

### Stepping Clauses

Stepping clauses place limits on the number of times the loop executes and increment *stepping variables* each time through the loop. More than one stepping clause may be specified in a single loop expression. In the case of multiple stepping clauses, the **`loop`** iteration stops as soon as any stepping clause terminates.

[Clause]

**`repeat`** *expr*

The repeat clause repeats the loop *expr* number of times.

Interaction 8-4. The repeat clause.

```
cm> (loop repeat 10 collect (random 100))
(9 3 93 58 23 0 79 48 27 60)
cm> (loop repeat (random 5) sum 10)
30
cm>
```

[Clause]

**`for`** *var* = *expr* `[then` *expr2*`]`

Sets variable *var* to the value of *expr* on each iteration. If **`then`** clause is supplied then *expr* is the initial value of *var* and *expr2* is its value on subsequent iterations.

Interaction 8-5. The for clause.

```
cm> (loop repeat 5 for x = (random 100) minimize x)
12
cm> (loop repeat 5 for x = -99 then (random 10)
          collect x)
(-99 4 8 6 4)
cm>
```

[Clause]

**`for`** *var* `[from` *low*`] to` *end* `[by` *step*`]`

Increments *var* until equal to *end* and then stops iteration. If **`from`** is not specified then *low* always defaults to 0. If **`by`** is not specified then *step* always defaults to 1.

Interaction 8-6. The for clause.

```
cm> (loop for i to 10 collect i)
(0 1 2 3 4 5 6 7 8 9 10)
cm> (loop for i from -1 to 2 by .5 collect i)
(-1 -0.5 0.0 0.5 1.0 1.5 2.0)
cm>
```

[Clause]

**`for`** *var* `[from` *low*`] below` *end* `[by` *step*`]`

Increments *var* while less than *end*.

Interaction 8-7. The below clause.

```
cm> (loop for i below 10 by 2 collect i)
(0 2 4 6 8)
cm>
```

[Clause]

**`for`** *var* `from` *top* `downto` *end* `[by` *step*`]`

Decrements *var* until equal to *end*.

Interaction 8-8. The downto clause.

```
cm> (loop for i from 0 downto -10 by 2 collect i)
(0 -2 -4 -6 -8 -10)
cm>
```

[Clause]

**`for`** *var* `from` *top* `above` *end* `[by` *step*`]`

Decrements *var* while greater than *end*.

Interaction 8-9. The above clause.

```
cm> (loop for i from 10 above 2 collect i)
(10 9 8 7 6 5 4 3 2 )
cm>
```

[Clause]

**`for`** *var* `in` *list* `[by` *fn*`]`

[Clause]

**`for`** *var* `on` *list* `[by` *fn*`]`

The **`in`** clause maps *var* over successive elements in *list*. An optional **`by`** function increments the *list*. The **`on`** clause maps successive tails of *list*.

Interaction 8-10. Example list iterations.

```
cm> (define test '(a b c))

cm> (loop for v in (reverse test) collect v)
(c b a)
cm> (loop for v in test collect (list v v))
((a a) (b b) (c c))
cm> (loop for v on test collect v)
((a b c) (b c) (c))
cm>
```

## Action Clauses

Action clauses implement iteration tasks and return values from the loop.

[Clause]
  **`do`** *expr ...*

Evaluates every *expr* on each iteration. Returns boolean false.

[Clause]
  **`collect`** *expr*
[Clause]
  **`append`** *expr*

The **`collect`** clause collects values of *expr* each iteration and returns the values in a list while the **`append`** clause appends the contents of *expr*, which must evaluate to a list.

[Clause]
  **`sum`** *expr*
[Clause]
  **`minimize`** *expr*
[Clause]
  **`maximize`** *expr*
[Clause]
  **`count`** *expr*
[Clause]
  **`thereis`** *expr*

Return the sum, minimal and maximal values of *expr* in the loop, respectively. The **`count`** clause returns the number of times *expr* is true and **`thereis`** returns stops iteration and returns true from the loop as soon *expr* is true.

Interaction 8-11. Example action clauses.

```
cm> (loop repeat 2 do (print 'hi))
hi
hi
#f
cm> (loop repeat 10 collect (random 50))
(43 42 27 18 26 20 44 43 23 30)
cm> (loop repeat 10 sum (random 50))
176
```

```
cm> (loop for i below 10 count (even? i))
5
cm> (loop repeat 10 minimize (random 50))
2
cm> (loop repeat 10 maximize (random 50))
43
cm> (loop for x below 10 thereis (= x -1))
#f
cm>
```

## Conditional Clauses

Conditional clauses place conditions on the evaluation of action clauses, or on how many times the **`loop`** will iterate.

[Clause]

**`when`** *test action* `[and ...]`

[Clause]

**`unless`** *test action* `[and ...]`

[Clause]

**`while`** *test ...*

[Clause]

**`until`** *test ...*

The **`when`** clause executes the *action* clause only if *test* is true. The optional **`and`** operator be used to make multiple actions dependent on a single test. The **`unless`** clause executes the *action* clause only if *test* is false. The **`while`** clause terminates iteration if *test* is false, and the **`until`** clause terminates if *test* is true.

Interaction 8-12. Example conditional clauses.

```
cm> (loop for i below 10 when (even? i) collect i)
(0 2 4 6 8)
cm> (loop for i below 10 unless (even? i) collect i)
(1 3 5 7 9)
cm> (loop for n = (random 10) until (even? n) collect n)
(9 3)
cm> (loop for n = (random 10) while (even? n) collect n)
()
cm> (loop for i below 10
          for j = (random 10)
          when (even? j) collect i and collect j)
(2 2 4 6 7 4 8 0 9 4)
cm>
```

## Initialization Clauses

[Clause]

**`with`** *var* `[=` *expr*`]` `[and ...]`

Defines *var* as a local loop variable. *var* is initialized to false unless it is explicitly set using the optional **`=`** operator. The optional **`and`** operator allows more than one variable to be defined by the **`with`** clause.

Interaction 8-13. Example initialization clause.

```
cm> (loop with hz = 220
          for h from 1 to 8
          collect (* hz h))
(220 440 660 880 1100 1320 1540 1760)
cm>
```

### Finalization Clauses

[Clause]

**`finally`** *expr*

[Function]

(**`return`** *expr*)

Evaluates *expr* after the iteration has stopped. The **`return`** function can be used anywhere inside **`loop`** to return *expr* as its value.

Interaction 8-14. Example finalization clause.

```
cm> (loop repeat 10
          for sum = 0 then (+ sum (random 100))
          finally (return (/ sum 10.0)))
48.6
cm>
```

## Advanced loop Features

The iteration possibilities afforded by **`loop`** are almost limitless. The following discussion points out a few loop features that involve more than one clause.

### Parallel clauses

More than one stepping clause or action clause can be defined in a single loop. In the case of multiple stepping clauses the first clause that terminates will stop the entire loop.

Interaction 8-15. Multiple stepping clauses.

```
cm> (loop for i below 4
          for j from 100 by -10
          for k = (random 100)
          collect (list i j k))
((0 100 1) (1 90 92) (2 80 81) (3 70 2))
cm> (loop for i below 4
          collect i
          collect (expt 2 i))
(0 1 1 2 2 4 3 8)
cm>
```

### Loops inside loops

A single loop may contain other loops inside it.

Interaction 8-16. A loop in a loop.

```
cm> (loop for i from 0 by 10 to 20
          collect (loop repeat 4 for j from i collect j))
((0 1 2 3) (10 11 12 13) (20 21 22 23))
cm>
```

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file loop.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 9 *Etudes*, Op. 3: Iteration, Rows and Sets

In *Etudes*, Op.3 we work through exercises that demonstrate the use of iteration in twelve-tone and set theory operations.

## Op. 3, No. 1: Converting MIDI Key Numbers to Pitch Classes

In this first exercise we implement **`list->pcs`**, a function that converts a list of MIDI key numbers into pitch classes. A pitch class can be thought of as the "remainder" of a key number after all octave information has been "subtracted out". We will represent pitch classes as integers between 0 and 11. There are only 12 pitch classes because there are only 12 divisions per octave in the standard scale. For example, all C key numbers are pitch class 0, all F-sharps and G-flats are pitch class 6, and so on.

The Lisp function **`mod`** can be used to convert a key number into a pitch class. The **`mod`** function returns the absolute value of the remainder from the division of two integers. We can use **`loop`** to see the effects of **`mod`** on a range of numbers:

Interaction 9-1. Using loop to iterate numbers mod 12.

```
cm> (loop for i from 60 to 72 collect (mod i 12))
(0 1 2 3 4 5 6 7 8 9 10 11 0)
cm> (loop for k from 0 downto -12 collect (mod k 12))
(0 1 2 3 4 5 6 7 8 9 10 11 0)
cm>
```

Now that we have seen how to iterate a series of numbers mod 12, defining the example function is very easy. The only difference is that the function will iterate over a list of numbers, rather than a range. We split our programming task into two small functions so that we can convert single key numbers or lists of key numbers using our specialized program names.

Example 9-1. Converting key numbers to pitch classes.

```
(define (keynum->pc k)
  (mod k 12))

(define (list->pcs knums)
  (loop for k in knums collect (keynum->pc k)))
```

**`list->pcs`** uses loop to iterate a variable *k* over every key number in a list of key numbers input into the function. The **`collect`** clause collects the results of **`keynum->pc`** into a new list and returns that list as the value of the loop. Since the loop expression is the last expression in the body of the function definition, the value returned by loop becomes the value of the function.

Interaction 9-2. Pitch classes from measure 15 of Berg's Violinkonzert.

```
cm> (define albans-notes '(55 58 62 66 69 72 76 80 83 85 87 89))

cm> (list->pcs albans-notes)
(7 10 2 6 9 0 4 8 11 1 3 5)
cm>
```

## Op. 3, No. 2: Normalizing Pitch Classes

Note that in Interaction 9-2 the series of pitch classes returned by **`list->pcs`** is close to, but not the same as, a twelve-tone row. To form a twelve-tone row we would need to convert the list of pitch classes into a list of intervals, where the first interval in the row is 0 — the “root” for row transpositions. We will take this opportunity to define a new function that normalizes a list of pitch classes or key numbers into zero based intervals:

Example 9-2. Normalizing a list of pitch classes or key numbers.

```
(define (normalize-pcs knums)
  (loop with root = (first knums)
        for k in knums
        collect (keynum->pc (- k root))))
```

In this function a loop shifts the values in the input list by subtracting out the amount of the initial value represented in the list. Recall that the **`with`** operator is used to initialize, rather than step, a looping variable. The expression **`with root = (first knums)`** directs loop to define a variable *root* and bind it to the first element in the input list before loop starts iterating. The key feature to remember about a **`with`** clause is that it sets variables just one time. A **`for`** clause, on the other hand, sets variables each time through the loop.

Interaction 9-3. Normalizing pitch classes.

```
cm> (define albans-notes
      '(55 58 62 66 69 72 76 80 83 85 87 89))

cm> (normalize-pcs (list->pcs albans-notes))
(0 3 7 11 2 5 9 1 4 6 8 10)
cm>
```

## Op. 3, No. 3: Matrix Operations

In this example we define four functions which implement matrix operations. The function **`retrograde-row`** returns the retrograde version of a row. The function **`transpose-row`** shifts the row to a new pitch class. The function **`invert-row`** inverts a twelve-tone row. The function **`row->matrix`** returns a list of lists that represents the twelve rows of a "Prime by Inversion" matrix.

Example 9-3. Twelve-Tone Functions.

```
(define (retrograde-row row)
  (reverse row))

(define (transpose-row row to)
  (loop for pc in row collect (keynum->pc (+ pc to))))

(define (invert-row row)
  (loop for pc in row collect (keynum->pc (- 12 pc))))

(define (retrograde-invert-row row)
  (retrograde-row (invert-row row)))

(define (row->matrix row)
  (loop for i in (invert-row row)
        collect (transpose-row row i)))
```

Interaction 9-4. Twelve-tone matrix operations.

```
cm> (define albans-row '(0 3 7 11 2 5 9 1 4 6 8 10))

cm> (retrograde-row albans-row)
(10 8 6 4 1 9 5 2 11 7 3 0)
cm> (transpose-row albans-row 3)
(3 6 10 2 5 8 0 4 7 9 11 1)
cm> (invert-row albans-row)
(0 9 5 1 10 7 3 11 8 6 4 2)
cm> (retrograde-invert-row albans-row)
(2 4 6 8 11 3 7 10 1 5 9 0)
cm> (pprint (row->matrix albans-row))
((0 3 7 11 2 5 9 1 4 6 8 10)
 (9 0 4 8 11 2 6 10 1 3 5 7)
 (5 8 0 4 7 10 2 6 9 11 1 3)
 (1 4 8 0 3 6 10 2 5 7 9 11)
 (10 1 5 9 0 3 7 11 2 4 6 8)
 (7 10 2 6 9 0 4 8 11 1 3 5)
 (3 6 10 2 5 8 0 4 7 9 11 1)
 (11 2 6 10 1 4 8 0 3 5 7 9)
 (8 11 3 7 10 1 5 9 0 2 4 6)
 (6 9 1 5 8 11 3 7 10 0 2 4)
 (4 7 11 3 6 9 1 5 8 10 0 2)
 (2 5 9 1 4 7 11 3 6 8 10 0))
cm>
```

In Interaction 9-4 the function **`pprint`** was used to "pretty print" the value returned by **`row->matrix`** so that the rows in the matrix are easy to read.

## Op. 3, No. 4: Determining Normal Order

The *normal order* of a group of notes is the most "tightly packed" permutation of its interval content. The most tightly packed permutation is the inversion of the set with the smallest total span. In the case of a "tie", the permutation with smallest intervals "packed leftward" in the set is the winner. For example there are 4 permutations of a dominant seventh chord:

Table 9-1. Inversions of a dominant seventh chord.

| inversion | intervals |
|---|---|
| root | (0 4 7 10) |
| first | (0 3 6 8) |
| second | (0 3 5 9) |
| third | (0 2 6 9) |

The set (0 3 6 8) is the normal order because this permutation spans only 8 semitones and is therefore the most compact.

The first step in determining the normal order of a group of notes is to define a function that accepts a list of key numbers and returns its pitch class content with no duplications. We will implement this behavior using a "scratch pad" list that represents the 12 possible pitch classes of an octave. Each of the twelve elements in the list stands for the pitch class whose number corresponds to the element's position in the list. So the 0'th element in the list corresponds to pitch class 0, the element at position 1 corresponds to pitch class 1, and so on. Each element in the scratch pad list is initially set to false. A false element means that a pitch class for that position has not yet been recorded in the list. A pitch class is "remembered" in the scratch pad by iterating over the input note list and setting the elements in the scratch pad to true for each pitch class in the input list. Thus, if a pitch class is not encountered in the input list the corresponding element in the scratch pad will remain false. But any pitch class in the note set that is encountered one or more times will have its element in the scratch pad set to true. Once every pitch class in the input list has been processed, a second loop iterates through the scratch pad list and returns only those positions in the scratch pad for which a pitch class was actually recorded.

Example 9-4. Determining pitch class sets.

```
(define (reduce-pcs knums)
  ;; create a scratch pad list with 12 locations to
  ;; remember the pitch classes in knums. each element
  ;; in the pad is initially false.
  (let ((pad (make-list 12)))
    ;; remember each pitch class in knums by setting
    ;; its position in the scratch pad to true.
    (loop for k in knums
          for pc = (keynum->pc k)
          do (list-set! pad pc true))
    ;; iterate over the scratch pad and return
    ;; a list of the positions where pitch classes
    ;; were recorded
    (loop for i in pad
          for j from 0
          when i collect j)))
```

The function **`reduce-pcs`** defines the local variable *pad* to be a "scratch pad list" of 12 elements. **`make-list`** automatically sets each element in this list to false. The body of **`reduce-pcs`** consists of two loops. The first loop iterates over the input list and, for each pitch class it encounters, sets the element that corresponds to that pitch class in the scratch pad to true. The loop doesn't return anything; its **`do`** clause simply invokes the function **`list-set!`** to set each element at the specified *pc* position to true.

The second loop in the body consists of three clauses. The first clause **`for i in pad`** steps the variable *i* over every value in **`pad`**. Since the pad contains only true or false values the *i* variable is set to true at positions where a pitch class was recorded, otherwise it is set to false. The second clause **`for j from 0`** simply steps the variable *j* in parallel with the position of *i* in the scratch pad. The third clause **`when i collect j`** collects the value of *j* only when the value of *i* is true. The loop therefore returns exactly those positions in the scratch pad at which a pitch class had been recorded. This second loop is the last form in the body of the function so its value is returned as the value of the function call.

Interaction 9-5. Calling reduce-pcs on two octaves of G dom7.

```
cm> (define g-dom7 '(55 59 62 65 67 71 74 77))

cm> (reduce-pcs g-dom7)
(2 5 7 11)
cm>
```

The *g-dom7* variable defines a G dominant 7th chord doubled over two octaves. **`reduce-pcs`** returns the list of pitch classes in that chord with all duplicates removed. (The pitch class 2 is D, 5 is F, 7 is G and 11 is B).

The next step is to define a function that *permutes*, or rotates, a list of pitch classes so that each of its inversions can be tested for compactness. To rotate the list to the next inversion means to move the element that is currently at the front of the list to the back of the list. In order to do this we first "destructure", or split, the list into two separate lists. The first will contain all but the first element, and the second will contains only the first element.

The Lisp function **`rest`** will return the first list we need:

Interaction 9-6. A list of all but the first element.

```
cm> (define test-set '(2 5 7 11))

cm> (rest test-set)
(5 7 11)
cm>
```

The second list must contain only one element: the pitch class that is currently at the front of the input list. The Lisp accessor **`first`** returns the first element of a list, the function **`list`** can then be used to turn that value into a list:

Interaction 9-7. A list of only the first element.

```
cm> (first test-set)
2
cm> (list (first test-set))
(2)
cm>
```

Given the two destructured lists in Interaction 9-7 and Interaction 9-6 we can use the Lisp function **`append`** to "glue" them back together in the order we want:

Interaction 9-8. Reordering the destructured lists.

```
cm> (append (rest test-set) (list (first test-set)))
(5 7 11 2)
cm>
```

We are now ready to implement our function to permute a pitch class set:

Example 9-5. Permute pitch class set.

```
(define (permute-set set)
  (let ((front (list (first set)))
        (others (rest set)))
    (append others front)))
```

Interaction 9-9. Examples of set permutation.

```
cm> (permute-set test-set)
(5 7 11 2)
cm> (permute-set (permute-set test-set))
(7 11 2 5)
cm>
```

The last helper function we need to define is **`set-weight`**, a function that returns a measure of how compact a set is. Rather than checking the span of each inversion and then checking "leftward packing" in case of a tie, we implement a single function that returns a "total measure" of the compactness of a set. To do this we will use two functions that have already been introduced: the Lisp function **`expt`** and our own function **`normalize-pcs`**. Our method is quite simple: to measure the compactness of a pitch class list we will first convert the list into a normalized (zero based) set and then treat each number in the set as exponents. The power of 2 value of each exponent (interval) will then be summed together to determine the total weight. By summing exponential values a single large interval will always weigh more than smaller ones added together.

Example 9-6. Set permutation.

```
(define (set-weight set)
  ;; set must be normalized (zero based)
  (loop for s in set sum (expt 2 s)))
```

Interaction 9-10. Comparing inversion weights of a dom7.

```
cm> (set-weight '(0 4 7 10))
1169
cm> (set-weight '(0 3 6 8))
329
cm> (set-weight '(0 3 5 9))
553
cm> (set-weight '(0 2 6 9))
581
cm>
```

The **set-weight** function correctly identifies (0 3 6 8) as the most compact rotation, i.e. the normal order, of the dominant 7th chord. We are now ready to implement the actual function:

Example 9-7. Computing the normal order of a note set.

```
(define (normal-order notes)
  (let ((pclist (reduce-pcs notes)))
    (loop with memory and winner
          repeat (length pclist)
          for normal = (normalize-pcs pclist)
          for weight = (set-weight normal)
          do
          (if (or (not winner)
                  (< weight memory))
            (begin (set! winner normal)
                   (set! memory weight)))
          (set! pclist (permute-set pclist))
          finally (return winner))))
```

The **normal-order** function first calls our helper **reduce-pcs** to convert the input list into a list of pitch classes with all duplicate key numbers removed. The main processing loop then iterates as many times as there are pitch classes in *pclist*. The *winner* variable will hold the permutation of *pclist* with the smallest weight, i.e. the normal order of the note list. The variable *memory* will be used to "remember" the smallest weight as each inversion of *pclist* is compared to the previous one. On each iteration of the loop the stepping variable *normal* is set to the normalized version of *pclist* and its weight is calculated by the **set-weight** function. The **do** action clause contains two Lisp expressions: an **if** conditional statement and a **set!** assignment statement. The purpose of the assignment statement **(set! pclist (permute-set pclist))** is to reset the *pclist* variable at the end of the loop so that when the loop returns back to the top the variable already contains the next inversion of the pitch class set to check. The purpose of the **if** statement is to establish the very first version of *pclist* as the "winner" unless a subsequent inversion can be shown to have a smaller set weight. The *then* clause of the **if** statement actually consists of two assignment statements that are "bundled together" into a single *then* expression using the **begin** function. (Without the surrounding begin expression the first **set!** expression would be interpreted as the *then* clause and the second **set!** expression would be considered to be an *else* clause!) The test expression: **(or (not winner) (< weight memory))** evaluates to true if either (1) the *winner* variable has not been set (i.e. it is still false from the initial **with** clause) or (2) the current value of *weight* is less than the value saved in *winner*. The **`finally`** clause executes at the very end of the loop, after every inversion of *pclist* has been checked. The **`finally`** clause uses the **`return`** function to return whatever version of *pclist* was saved in *winner*. Since the loop is the last statement in the body of the function, the value that the loop returns becomes the value that the function returns.

Interaction 9-11. Computing normal orders.

```
cm> (normal-order g-dom7)
(0 3 6 8)
cm> (normal-order '(72 64 67))
(0 4 7)
cm> (normal-order '(65 64 75 70))
(0 1 2 7)
cm>
```

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file sets.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 10 Parameterized Sound Description

Of all noises, I think music is the least disagreeable.

— Samuel Johnson

The introduction to this book began with a discussion of three representation levels, or layers, that music composition applications commonly address. Applications that operate at the *acoustic level* record and output sound based on a digitized representation of an acoustic waveform. Computer applications in the *score level* use a symbolic encoding to represent specific attributes, or qualities, of sound. In the *metalevel*, compositional algorithms, processes and relationships are expressed. Each of these levels is a layer of abstraction that encapsulates, or hides, details that are explicit in the next lower level. Figure 1 shows a single musical example depicted at the three abstraction levels. Each depiction is represented as a file containing an encoding of the example and an estimate of its size in bytes. The score and acoustic representations in Table 2 and 3 are generated automatically from the single metalevel construct in Table 1.

Table 10-1. Metalevel representation of a self-similar process.

sier.cm: 8 lines, 225 bytes.

```
(define (sierpinski knum ints dur depth)
  (process for i in ints
           ;; output new sound event
           output (new midi :time (now) :keynum (+ knum i)
                        :duration dur :channel depth)
           when (> depth 1)
           sprout (sierpinski (+ knum i 12) ints len
                              (/ dur 3) amp (- depth 1))
           wait dur))
```

Table 10-2. MIDI and Csound score level sound events from Table 1.

```
sier.midi: 250 messages, 1000 bytes.   sier.sco :117 lines, 4200 bytes.
 0 <Note-On 0 25 63>                   i3 0.000 1.000 34.647;
 0 <Note-On 0 37 63>                   i2 0.000 0.333 69.295;
 0 <Note-On 0 49 63>                   i1 0.000 0.111 138.591;
53 <Note-Off 0 49 127>                 i1 0.111 0.111 207.652;
 0 <Note-On 0 56 63>                   i1 0.222 0.111 184.997;
53 <Note-Off 0 56 127>                 i2 0.333 0.333 103.826;
 0 <Note-On 0 54 63>                   i1 0.333 0.111 207.652;
53 <Note-Off 0 37 127>...              i1 0.444 0.111 311.126;...
```

Table 10-3. Acoustic level waveform samples from Table 2.

sier.snd: 350,000 samples, 700,000 bytes.

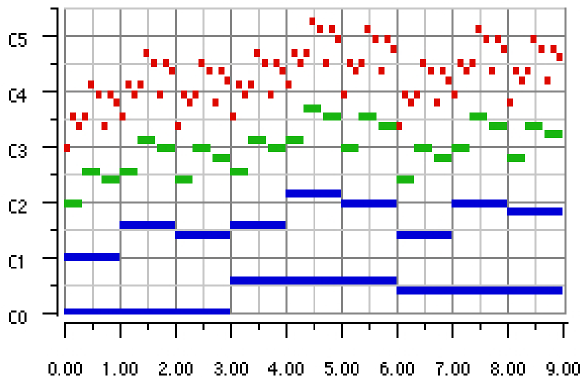
```
732e 646e 0000 1c00 0000 0000 0000 0300 0000 2256
0000 0100 0000 0000 d0ff a0ff 75ff 44ff 0cff e9fe
c9fe 74fe 27fe 8dfe b1ff b100 ef00 d100 e300 0c01
1201 0c01 1c01 3f01 6801 7301 3f01 2701 c301 ff02
3504 1b05 b605 cf05 8305 9705 3b06 9006 5b06 8d06
1007 4106 d803 ff01 2d02 1803 df02 3d01 fcfe 15fd
73fc 03fd 6ffd fdfc 91fc c6fc e5fc 96fc 8bfc ...
```

This translation is accomplished by describing sound symbolically, using an abstraction called a *parameterized sound event*. A sound event is a general description of sound that can generate different score level encodings of the same data. There are many applications in the score level that overlap in functionality but differ in the exact data format they use to encode sonic information. Parameterized sound description allows the composer to represent all sound in a consistent, uniform manner while still being able to address the various applications and languages that operate at the score and synthesis composition levels.

## Parameterized Sound Events

A parameterized sound event encodes sound as a collection of properties called *sound parameters*. A sound is encoded by first defining an appropriate set of parameters to describe its characteristics and assigning these parameters specific values that constitute each instance of a sound. All sounds have a basic set of properties that event descriptions encode, but the complete set of parameters for a given sound event will depend both on the nature of the sound as well as on the score level synthesis algorithm that is used to produce its acoustic waveform. A useful way to think about sound parameters is that they define dimensions of compositional interest. Coordinates for each "sound point" are expressed by values along each parameter's axis. The job for the composer, then, is to articulate sound points in a multi-dimensional parameter space in musically satisfying ways. Many score level applications display sound points in graphical plots of parameterized sound data. Notation programs map event parameters onto the traditional five-line staff of Common Practice Notation. Many MIDI sequencers provide an alternate display style called “piano roll notation” that maps two or more sound parameters onto a two dimensional grid (Figure 10-1).

Figure 10-1. Piano roll graph displaying sound events from Figure 1a. The graph displays four sound parameters: the horizontal axis graphs start-time and duration, the vertical axis graphs frequency, and color is assigned to timbre (channel).



The remaining chapters in this book present various metalevel techniques that ultimately output events to produce patterns in parameterized sound data. We begin this learning process by first defining some basic properties of sound that all events encode and then discussing some common formats for expressing their parameter values.

## The Properties of Sound

When a sound source vibrates its oscillations disturb molecules in the surrounding medium. The elastic nature of the medium causes a chain reaction to occur as the disturbance is propagated from molecule to molecule. The result is a series of *waves* that radiate outward from the sound source in all directions. These sound waves consist of compressions and rarefaction of molecules in the medium brought about by the back and forth motion of the sound source. When molecules are pushed together (compression) the pressure in that vicinity of the medium is higher than in regions still at equilibrium. Similarly, when vibration in the opposite direction pulls molecules apart (rarefaction) the pressure at that locality is less than at points in the medium still at equilibrium. This smooth, continuous change in pressure on either side of an equilibrium point has a shape that can be graphed over time (Figure 10-2).

Figure 10-2. The scatter plot (blue) depicts the density of molecules in the medium at an instant in time. The line graph (black) depicts pressure as a function of time. Equilibrium, compression, and rarefaction are labeled E, C and R, respectively.

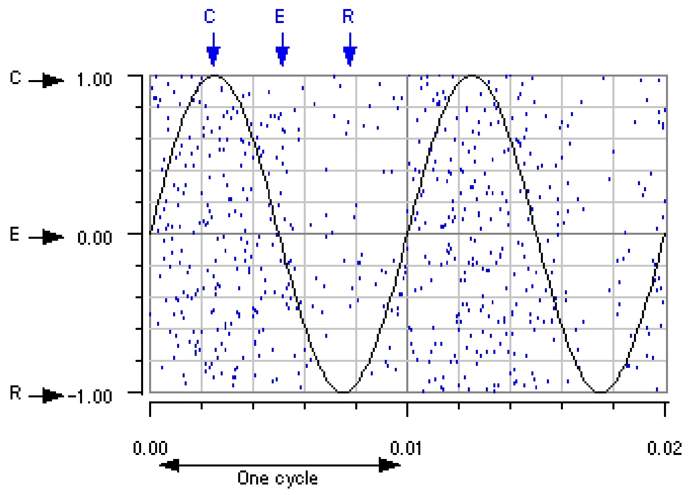
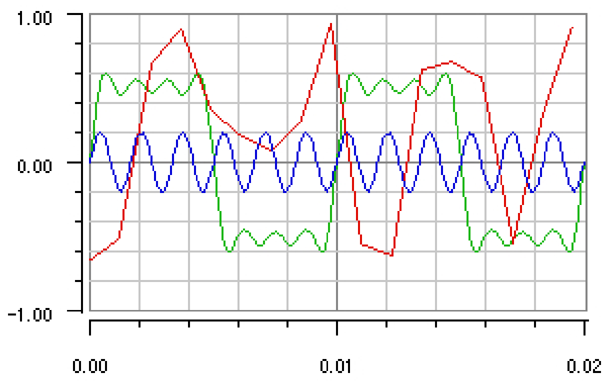


Figure 10-2 plots a waveform as a change in pressure over time. The vertical axis displays the *amplitude* of the waveform, a measurement of the amount of energy transmitted in its compressions and rarefaction. The waveform in Figure 2 can be seen to oscillate in a very regular way. The unit of regularity in a waveform like this is called a *cycle*, and the time a cycle takes to occur is called a *period*. A wave that oscillates at a completely uniform rate such as the one in Figure 2 is called a *sine wave* and the rate of oscillation of a sine wave is called *frequency*. Frequency is typically measured in *Hertz*, the number of cycles per second the waveform produces. Sine waves are the basic building blocks of all sound. Most musical instruments generate *complex waveforms* rather than sine waves. A complex waveform oscillates at multiple rates, or frequencies, at the same time (Figure 3). When these component frequencies are related by integer ratio, the sound is said to be *harmonic. Harmonic sound is generally perceived as a single fundamental frequency with an associated timbre* or tone color. Most, but not all, musical instruments produce harmonic sound. In contrast, *aperiodic* waveforms, or noise, have little or no discernible regularity in their vibration.

Figure 10-3. Plots of three different types of waves. A sine wave (blue, small amplitude), a complex waveform containing odd harmonics (green, medium amplitude), and noise (red, large amplitude).



## Sound Properties and Event Parameters

The preceding discussion of sound waves identifies some fundamental properties of sound. All sounds are comprised of one or more *frequency* components. The mixture of these components imparts a characteristic *timbre*, or tone color to the sound. Sound transmits a certain amount of *amplitude*, or energy, to our ears. All sounds also *start* and last for some *duration* of time. These five basic properties: start, duration, frequency, amplitude, and timbre are present either implicitly or explicitly in all parameterized sound event descriptions, including the symbols of traditional music notation. Of course additional sound properties could be added to the list, and properties may not correspond in a one-to-one manner with sound event parameters . For example, natural sounds have a location associated with them but this property may or may not be represented in a particular sound event (Table 1 and 2). The property of timbre, on the other hand, is part of every sound but it is so complex that its specification may be spread over several parameters Table 10-5, or only implicitly represented by the choice of instrument that renders the sound event description (Table 10-4).

Table 10-4. Sound event parameters for MIDI.

| **parameter** | **property** | **value** |
|---|---|---|
| time | time | 0.0 |
| duration | time | 0.5 |
| keynum | frequency | 60 |
| amplitude | amplitude | 64 |
| channel | timbre | 0 |

Table 10-5. Sound event parameters for FM synthesis.

| **parameter** | **property** | **value** |
|---|---|---|
| time | time | 0.0 |
| duration | time | 3 |
| frequency | frequency | 440 |
| amplitude | amplitude | 0.2 |
| amplitude-env | amplitude | (1 0 .25 .1 1 0) |
| mratio | timbre | 2 |
| index | timbre | 4 |
| index-env | timbre | (0 0 .5 1 1 0) |
| degree | amplitude | 0 |
| distance | amplitude | 1 |
| reverb | timbre | .01 |

## Parameter Data Formats

When viewed from the perspective of sound events, the activity of music composition consists of projecting musically interesting patterns into data assigned to event parameters. But we have seen that a single sound property may be expressed differently in two sound events. Moreover, each sound event may require its parameters to be described in a particular way, by using specific formats for data. It is important, then, that data description at the metalevel be flexible and interchangeable so that the composer can work with whatever format is most appropriate for a given situation. Chapter 2 presented the basic Lisp data types and discussed their possible musical relevance. All sound parameters values consist of specific interpretations of these data types. The remaining sections of this chapter present basic musical formats for their interpretation.

### Frequency Formats

Over the course of centuries Western composers have developed many systems that organize and relate frequency information. Some systems define relationships between absolute frequency values while others describe frequency in terms of relative distance, or motion. In computer composition, the implementation of a specific score level composition application may bring an additional constraint on the format a composer uses to express frequency data. For example, the MIDI protocol encodes frequency information as integer "key numbers". Notation programs, on the other hand, represent frequency by a symbolic language, while sound synthesis languages such as Common Lisp Music and Csound typically expect frequency to be encoded as raw Hertz values. Because there are so many ways to work with frequency, Common Music defines four interchangeable formats for its expression:

- in *Hertz*, or cycles per second
- as *keynum* positions in a scale
- as *note* names in a scale
- as *interval* distances between notes or key numbers

A *Hertz* value specifies frequency as the number of cycles per second of its waveform. Many synthesis languages expect frequency in Hertz because this format does not require an underlying tuning system or mode for its interpretation. In contrast to Hertz, the note and keynum formats *convert* to a frequency given the existence of some underlying scale that quantizes frequency space into discrete steps. The frequency of each scale degree depends on the tuning system that the scale employs. For example, the standard chromatic scale uses an equal tempered tuning system based on a C at 8.175 Hz with an interval of $2^{1/12}$ (100 cents) between each scale degree.

A *keynum* specifies the scale degree position of a frequency in a scale. Since a keynum is a numerical quantity like Hertz it can be used in arithmetic calculations. But unlike Hertz, keynums also encode pitch class and octave information and allows intervals to be calculated using addition and subtraction rather than multiplication and division. The MIDI protocol uses integer key numbers to identify specific "tones" that a Note-On message activates when it is sent to a MIDI synthesizer. Common Music extends the idea of key number to include *floating point key numbers*, such as 60.5 or 45.678, so that frequencies "in between" adjacent scale degrees in a tuning system can be specified. A floating point key number is interpreted *kkk.cc* where *kkk* is the integer key portion of the keynum and the fractional portion *.cc* is interpreted as *cc* cents above the Hertz value of *kkk*.

A *note* is a Lisp symbol that names a scale degree. Although note names vary from scale to scale as defined by the composer, a note name generally consists of a basic class name such as **`a`** or **`nem`**, an (optional) accidental sign and an octave number. For example, **`a4`** is the note name for 440 Hz in the standard chromatic scale, **`cs5`** is C-sharp in the fifth octave, and **`nem0`** is the first note in the first octave of the Pelog scale defined in Chapter 15. Note names are the most generally useful frequency format because they encode information about the spelling of a frequency in addition to its Hertz frequency, pitch class and octave register.

An *interval* is an integer that specifies the distance between two scale degrees. This distance can be expressed as a positive or negative number of steps between degrees or as a special *typed interval* that encodes information about an interval's spelling in addition to its size. See the Common Music dictionary for more information about typed intervals. In a given compositional context there may be one frequency representation that has advantages over the others. The following functions provide a mapping between the frequency formats discussed above.

[Function]
```
(hertz f [:in scale])
```
[Function]
```
(keynum f [:in scale] [:from scale] [:to scale])
```
[Function]
```
(note f [:in scale] [:from scale] [:to scale] [:accidental acc])
```
[Function]
```
(transpose f [int] [scale])
```

The functions **`hertz`**, **`keynum`**, and **`note`** convert a frequency specification *f* to the format for which the function is named. The function **`transpose`** shifts *f* by *int* intervals while preserving the current frequency format of *f*. All four functions operate with respect to a tuning system or mode optionally specified to the function. If *scale* is not specified the functions will use the default scale defined in the system's global **`*scale*`** variable. By default, this variable is defined to hold the standard chromatic scale, whose properties are discussed in the next section. The **`hertz`** function returns the Hertz frequency of a scale degree or note name:

Interaction 10-1. Converting to Hertz in the standard chromatic scale.

```
cm> *scale*
#<tuning "chromatic-scale">
cm> (hertz 'a4)
440.0
cm> (hertz 69)
440.0
cm> (hertz '(a4 c5 e))
(440.0 523.251 659.255)
cm>
```

The **`hertz`** function returns the key number of a Hertz value or note name. Hertz values always produce floating point key numbers, which encode microtonal inflection as a non-zero remainder.

Interaction 10-2. Converting to key numbers.

```
cm> (keynum 'a4)
69
cm> (keynum 440 :hz)
69.0
cm> (keynum '(440 550 660) :hz)
(69.0 72.863 76.019)
cm>
```

The **`note`** function returns the note name of a key number or Hertz value.

Interaction 10-3. Converting to notes.

```
cm> (note 69)
a4
cm> (note 440 :hz)
a4
cm> (note '(69 72 76))
(a4 c5 e5)
cm>
```

The **`transpose`** function shifts notes or key numbers by an interval amount. If a note and typed interval are specified then the transposition will reflect the interval's spelling.

Interaction 10-4. Transposing notes and key numbers.

```
cm> (transpose 69 6)
75
cm> (transpose 'a4 6)
ef5
cm> (transpose 'a4 (interval 'aug 4))
ds5
cm> (transpose '(a4 c5 e) (interval 'aug -2))
```

```
(gf4 bff4 df5)
cm>
```

These examples point out several features common to all four functions:

- In order to distinguish a Hertz number from a keynum number a Hertz value must be "tagged" by the keyword **`:hertz`** or **`:hz`**.
- Lists may be specified as well as single values. Lists allow aggregate frequency structure such as sets, rows and chords to be processed as a single operation.
- A note in a note list may provide an octave number or not. If not, its octave number will default to the last specified octave number in the list, or the 4th octave (middle-C octave) if no number is specified.

## The Standard Chromatic Scale

The frequency functions documented in the previous section all operate with respect to an underlying tuning system or mode. If no mode or tuning is specified then the functions use the system's default tuning system stored in the global variable **`*scale*`**. This scale is initially set to the standard chromatic scale but can be reset by the user. The standard chromatic scale is implemented with 11 octaves of note entries. The first octave in the scale is numbered -1; the Middle-C octave is 4 and the highest octave number is 10. Middle-C (written C4) is keynum 60 and has a frequency of 261.625 Hz. There is no upper octave limit for key numbers or Hertz values.

## Note Names in the Standard Chromatic Scale

A note in the standard chromatic scale consists of a pitch class letter, an optional accidental and an octave number:

Table 10-6. Note spellings in the standard chromatic scale.

| name | accidental | octave |
|---|---|---|
| c, d, e, f, g, a, b | ff, f, n, s, ss | -1 to 10 |

Note entries include all natural (n), sharp (s), flat (f), double-sharp (ss) and double-flat (ff) spellings. The symbol **`r`** means rest.

Interaction 10-5. Note names in the chromatic scale.

```
cm> (keynum 'c4)
60
cm> (keynum 'dff4)
60
cm> (keynum 'bs3)
60
cm> (note 'c4 :accidental 's)
bs3
cm> (note 261.625 :hz :accidental 'ff)
dff4
cm> (keynum 'c-1)
0
cm> (hertz 'b10)
31608.531
cm>
```

### Key numbers in the Standard Chromatic Scale

Key numbers in the standard chromatic scale may be expressed as integers or a floating point numbers. A floating point keynum is interpreted *kkk.cc* where *kkk* is the integer keynum and *cc* is cents above the Hertz value of *kkk*. For example, 60.5 means 50 cents (quarter tone) above the frequency of keynum 60, or 269.291 Hz.

Interaction 10-6. Key numbers in the chromatic scale.

```
cm> (keynum 439 :hz)
68.96
cm> (hertz 69)
440.0
cm> (hertz 69.5)
452.892
cm> (note 69.5)
bf4
cm>
```

### Intervals in the Standard Chromatic Scale

Intervals measure distances between scale degrees. In the standard chromatic scale this measurement can be expressed in terms of simple integer half steps or by using a specially encoded *typed interval* that includes the quality and spelling of an interval along with its size.

## Time Formats

A number of time formats are commonly used in computer composition: seconds, milliseconds, "ticks", and metric proportions scaled by a tempo factor. The basic format for time in Common Music is seconds. Time values can be expressed as Lisp integers, ratios and floats. The function **`rhythm`** to convert proportional and symbolic metric values to time in seconds.

[Function]

(**`rhythm`** *rhy* [*tempo*] [*beat*])

The **`rhythm`** function converts a metric value *rhy* to time in seconds. *rhy* can be a ratio, a rhythmic symbol (Table 10-7) or a list of the same. *Tempo* provides the optional metronome speed and defaults to 60. *Beat* specifies the metronome's pulse and defaults to .25, or quarter note. Global default metronome and beat values can be established by setting the system's **`*tempo*`** and **`*beat*`** variables, respectively.

Table 10-7. Table of Metric Symbols.

| *letter* | value | *letter* | value |
|---|---|---|---|
| **m** | maxima | **q** | quarter |
| **l** | longa | **e** | eighth |
| **b** | brevis | **s** | sixteenth |
| **w** | whole | **t** | thirty-second |
| **h** | half | **x** | sixty-fourth |

A rhythmic symbol consists of a metric letter from Table 10-7 that may be optionally preceded by a division letter: **t** for triplet or **q** for quintuplet, and optionally followed by any number of dots. Rhythmic symbols also support addition, subtraction and multiplication. The following table displays some representative rhythmic symbols.

Table 10-8. Examples of rhythmic symbols.

| | |
|---|---|
| quarter | **`q`** |
| triplet quarter | **`tq`** |
| dotted-eight | **`e.`** |
| triplet dotted sixteenth | **`ts.`** |
| triple dotted half | **`h...`** |
| quintuplet whole | **`qw`** |
| sixteenth plus triplet quarter | **`s+tq`** |
| whole minus triplet sixteenth | **`w-ts`** |
| whole times 4 | **`w*4`** |

Interaction 10-7. Examples of using the rhythm function.

```
cm> (rhythm 'q)
1.0
cm> (rhythm 1/4 120.0)
.5
cm> (rhythm 'w+q )
5.0
cm> (define *tempo* 120.0)

cm> (rhythm '(q th e.. x) )
(0.5 0.6666666666666666 0.4375 0.03125)
cm>
```

## Amplitude Formats

Amplitude is a particularly complex sound property, partly because the word is used with different meanings (loudness, intensity, pressure, and so on) and partly because its perception depends on a number of factors. These factors include the frequency of the sound (the Fletcher-Munson curves) as well as the listener's distance from the sound source. When computers and amplifiers are added to the equation the situation becomes even more complex. Since amplitude can be electronically boosted or reduced at any number of points along a signal's path the amplitude level that reaches our ears may have little to do with the amplitude value specified in the computer score. Acoustic amplitude is typically measured by pressure or terms of a *sound intensity level* (SIL), or Watts per meter squared. Intensity levels are defined from $10^{-12}$, the threshold of hearing, to $10^{0}$, the threshold of pain. Since 12 orders of magnitude is an enormous range of values, SIL are typically plotted in a logarithmic scale called the *Decibel scale* (dB). But dB values do not say how "loud" a sound is perceived, which is generally what composers are interested in. Composers typically specify amplitude in terms of a "psychological scale" using symbolic *dynamic levels* whose meaning corresponds only very loosely to the measured intensity levels in Table 10-9.

Table 10-9. Comparison of dynamic scale to SIL. (Backus)

| Dynamic level | Intensity (w/m2) | I/Ir | SIL (dB) |
|---|---|---|---|
| ffff (pain) | $10^{0}$ | $10^{12}$ | 120 |
| fff | $10^{-2}$ | $10^{10}$ | 100 |
| ff | $10^{-3}$ | $10^{9}$ | 90 |
| f | $10^{-4}$ | $10^{8}$ | 80 |
| mf | $10^{-5}$ | $10^{7}$ | 70 |
| mp | $10^{-6}$ | $10^{6}$ | 60 |

| | | | |
|---|---|---|---|
| p | $10^{-7}$ | $10^{5}$ | 50 |
| pp | $10^{-8}$ | $10^{4}$ | 40 |
| ppp | $10^{-9}$ | $10^{3}$ | 30 |
| pppp (threshold Ir) | $10^{-12}$ | $10^{0}$ | 0 |

Since amplitude measurement is so variable, synthesis languages often represent amplitude as a *normalized* value between 0.0 and 1.0, where 0.0 means silence and 1.0 means a fully saturated signal. The MIDI protocol represents amplitude values using a "key velocity" scale defined between 0 and 127, where 0 is silence and 127 is maximum. Most of the examples in this book use the normalized representation of amplitude (0.0-1.0) so that amplitude values may be easily computed or controlled by envelopes and other scaling operations. Composers who would like to work with the musical dynamic scale can use Common Music's **`amplitude`** function to convert symbolic amplitude levels into numerical values within a specified range:

[Function]

```
(amplitude amp [softest] [loudest] [power])
```

This function converts the logical amplitude *amp* into a value between *softest* and *loudest* according to the *power* curve. A logical amplitude is a number between 0.0 and 1.0 or one of the amplitude symbols shown in Table 10-10:

Table 10-10. Table of logical amplitude levels.

| | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| 0.0 | 0.1 | 0.2 | 0.3 | 0.4 | 0.5 | 0.6 | 0.7 | 0.8 | 0.9 | 1.0 |
| **niente** | **pppp** | **ppp** | **pp** | **p** | **mp** | **mf** | **f** | **ff** | **fff** | **ffff** |

The values of *softest,*

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# 11 Metastructure for Composition

In Chapter 3 we learned that the property list is one way to represent the characteristics of an object we want to model. Example 11-1 shows how two different sounds might be represented as property lists.

Example 11-1. Tone descriptions represented as property lists.

```
(define piano-tone '(pitch 440 duration 1/4 loudness mf))

(define viola-tone '(pitch 440 duration 1/4 loudness mf))
```

Assume for the moment that the composer would like to define a function called **`play-tones`** to generate sound from descriptions such as those shown in Example 11-1. To the composer the two definitions represent very different things. But from Lisp's point of view the two lists contain the exact same information and are literally equal to one another:

Interaction 11-1. Lisp cannot distinguish between the two tones.

```
cm> (equal? piano-tone viola-tone)
#t
cm>
```

Since both lists contain the same information and neither list declares what *type* of sound it represents, it will be impossible for **`play-tones`** to produce a piano tone from the first description and a viola tone from the second. One possible solution to this problem might be to add a *type symbol* to the front of each list that "declares" what type of tone each list is:

Example 11-2. Adding type symbols to the front of each sound description.

```
(define sounds
  '((piano-tone pitch 440 duration 1/4 loudness mf )
    (viola-tone pitch 440 duration 1/4 loudness mf )
    (metal-crash duration 2 loudness ff)))
```

Given a list of sound descriptions such as those shown in Example 11-2 the composer could then design the **`play-tones`** function to check the symbol in the first element of each tone description to determine the action appropriate to the sound that the symbol describes. The addition of a type symbol to the front of each list is a primitive form of *application datatyping*, in which a programmer "declares" the type and format of data that will be processed by the application. Adding type symbols to property lists was actually used by early Lisp researchers as a primitive form of classifying the data that was being processed by their programs. Most computer programs that model the "real world" or assist human activity in some way are faced with the need to represent various objects and behavior that make up the real world application. The classification scheme used by the early researchers was only a very primitive method for doing this. It is not adequate for several reasons:

- A classification based on type symbols is expressively very weak. While it is possible to have piano-tones and viola-tones, there is no way to express the fact that both tones are *specializations* of a more general type of data called a Tone that is distinguishable from, say, other data that classified as Noise. The more general types Tone and Noise are completely absent from the representation.
- For each new type of Tone, the composer is required to add more conditional checks and type-specific behavior to the **`play-tones`** function. This function will soon become an unmanageable programming task as the number of different sounds in the composition increases.

- To the underlying language the data are still just lists. it is not possible to *optimize* the representation in any way.

During the 1980's Lisp adopted a powerful new methodology for modeling real world applications. This methodology is called *object-oriented programming*. In object oriented programming a domain of interest is modeled, or represented, by defining the application's data and behavior in terms of *classes* and *methods* that belong to an *object system*. An object system allows the state and behavior of the objects it represents to be classified and described. The application datatypes defined in the object system are efficiently implemented and fully integrated into Lisp's own hierarchy of datatypes. Common Music is an example of a software system that represents its data and behavior using an object system. In the case of Common Music, the application domain is musical composition and the object modeling involves the representation of compositional structure and its behavior during sound generation. Before turning our attention to the specifics of Common Music's representation, we first look at some general characteristics of object oriented design.

## Objects, Classes, Instances and Slots

An object system is a mechanism for defining the characteristics of data and the behavior of functions in an application program. An *object* is a software unit that represents something about the domain being modeled. Most object systems distinguish between two types of objects. A *class* is an object that describes the general characteristics of the data being modeled. These characteristics are sometimes referred to as *properties*, *slots* or *instance variables*. We will always use the term slot to refer to a defined characteristic of a class or instance. When a class is defined it can be declared to be a *subclass*, or specialization, of another, more general, class called a *superclass*. The superclass-to-subclass relationship means that every slot defined in the superclass is *inherited*, or automatically present, in the subclass. A subclass may also declare additional slots not found in the superclass. An *instance* is a specific, unique member of a class. Instance objects are the data that an object-oriented program manipulates.

## Creating Instances

Every instance that a program manipulates is *instantiated*, or created, from some class description. When an instance is created it receives all of the slots defined by the class together with whatever *default values* the class defines for those slots. A slot's default value is overridden if the user specifies some other value for that slot when the instance is created. We will always create instances by calling the special macro function **`new`**:

[Macro]

(**`new`** *class* [*init value*] *...*)

**`new`** creates an instance from a specification of its class and initial slot values. *Class* is the symbol class name of the instance to create. Any number of *slot initializations* may be specified after the class. Each slot initialization is a pair: a keyword slot name followed by its value in the new instance. A keyword is a symbol that is prefixed with a colon. For example, to create an instance of the **`midi`** event class with a time value of .5 would look like:

```
(new midi :time .5)
```

where **`midi`** is the name of the class, **`:time`** is the keyword form of the slot named **`time`** and .5 is its value in the new instance.

Interaction 11-2. Creating midi instances with new.

```
cm> (new midi)
```

```
#i(midi keynum 60 duration 0.5 amplitude 64 channel 0)
cm> (new midi :keynum 90 :time .5 :duration 3)
#i(midi time .5 keynum 3 duration 3 amplitude 64 channel 0)
cm>
```

Common Music normally prints sound events using a specially formatted display:

**#i(*class slot value ...*)**

The special prefix **#i** tells the user that an instance is being shown. The instance is displayed as a list containing the name of the instance's class followed by all its slots and values. This printing feature is a convenient way to "inspect" the various characteristics of a sound. The **#i** prefix tells you that you are looking at an instance object, not a list.

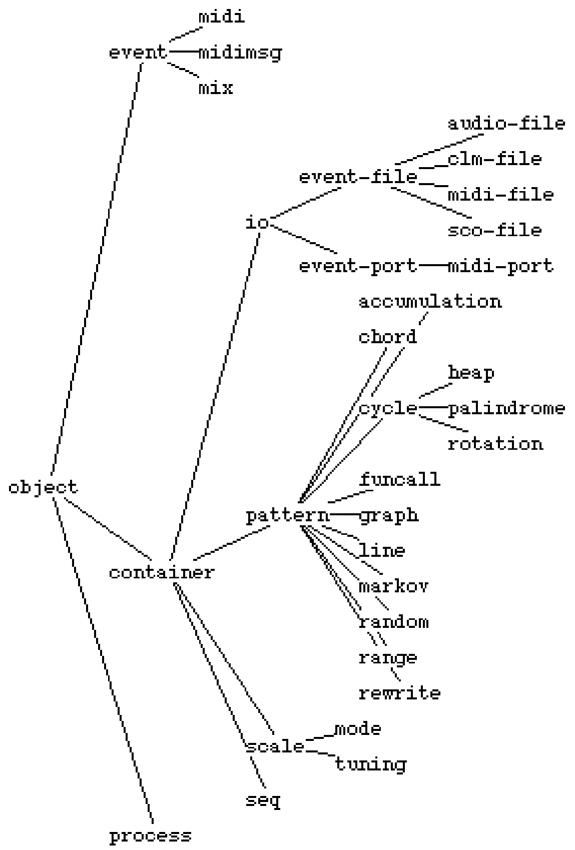
Lisp Note:
An **#i** expression may appear in upper case letters and the order of slots may vary in different Lisp implementations.

## The Common Music Class Taxonomy

There are three broad classifications of objects within the hierarchy defined by Common Music:

- Events define the characteristics of musical sounds that the composer works with.
- Containers aggregate musical data into larger units of structure.
- Processes implement procedural description of music.

Figure 11-1. A graph of Common Music's superclass->subclass hierarchy.



## Event Classes

An event is an object that represents a sound of some kind. Common Music allows the composer complete control over the definition of sound events but several specialized classes have already been defined for use with this book. The **`midi`** event class is the sound description that we will be concerned with in the next few chapters.

### The midi Event Class

The **`midi`** event is a sound representation for working with MIDI synthesizers, sequencing programs, and notation programs such as Finale and Sibelius that are able to notate MIDI data in traditional score format.

**`midi`** supports the following slot initializations:

:time *number*
    Sets the start time of the event in the score.
:keynum {*keynum* | *note*}
    Frequency expressed as a keynum or note. (See Chapter 14 "Tunings and Scales")
:duration *number*
    The duration of the event in seconds.
:amplitude *number*
    A logical amplitude 0.0 < 1.0 or an integer velocity 0 to 127. Defaults to 64.
:channel *integer*
    A MIDI channel number 0-15. Defaults to channel 0.

## Container Classes

Container classes support the aggregation of musical material. The **`seq`**, **`pattern`**, and **`io`** classes all implement different strategies for managing the objects they contain.

### The seq Container Class

A **`seq`** is a nameable object that implements musical sequencing. A seq is a nameable object that facilitates the sequencing of musical objects. These objects are stored as a time sorted list of *subobjects*. The subobject list may contain events, processes, and other sequences. When sequences are generated the start times of the subobjects are always calculated relative to the start time of the seq itself.

**`seq`** supports the following slot initializations:

:time *number*
    Sets the start time of the midi event in the score.
:name {*symbol* | *string*}
    Sets then name of the object to the *symbol* or *string*.
:subobjects *list*
    A list of subobjects.

Interaction 11-3. Creating a seq object.

```
cm> (new seq :name 'test :time 10
         :subobjects
         (loop for i below 4
               collect (new midi :time i)))
#<seq: "test">
cm> (object-time #&test)
10
cm> (object-name #&test)
"stuff"
cm> (subobjects #&test)
(#i(midi time 0 keynum 60 duration 0.5 amplitude 64 channel 0)
 #i(midi time 1 keynum 60 duration 0.5 amplitude 64 channel 0)
 #i(midi time 2 keynum 60 duration 0.5 amplitude 64 channel 0)
 #i(midi time 3 keynum 60 duration 0.5 amplitude 64 channel 0))
cm>
```

Common Music defines a specialized printing method for containers that shows the name of the container when the object is printed in the Listener window. A container that has no name is displayed according to whatever method the Lisp implementation provides.

The special prefix **`#&`** "fetches" a container given its name. This feature saves the composer from having to define a variable just to store the object when it is created.

There are a number of accessor functions that are appropriate to use with events and containers. Consult the entry for **`seq`** in the Common Music Dictionary for more information.

## IO Classes

The **`io`** class is the superclass for musical scores. Most scores are stored as different types of files on the computer's hard drive. Functions that perform input or output to scores accept either **`io`** instances or strings that identify their file on the computer. Filenames that are specified to input or output functions must always be specified with a file name extension. This extension determines the type of IO instance that will created. Common Music defines scores class for the following file extensions.

Table 11-1. File extensions and score types.

| **extensions** | **score class** |
|---|---|
| .midi, .mid | `midi-file` for MIDI |
| .sco | `sco-file`for Csound |
| .aiff, .wav, .snd | `audio-file` for Common Lisp Music |
| .clm | `clm-file` for Common Lisp Music |
| .cmn, .eps | `cmn-file` for Common Music Notation |

## The midi-file Score

The **`midi-file`** score supports output to Level 0 MIDI files and input from either Level 0 or Level 1 MIDI files. Level 0 means that the data in the MIDI file constitutes a single *track*, or time line of MIDI events. Level 1 MIDI files contain multiple tracks of data. The **`midi-file`** class is automatically chosen when you specify a file with a ".midi" or ".mid" extension.

**`midi-file`** supports the following slot initializations:

:versions {*number* | *true*}
: Causes a new version of the file to be created each time output occurs. The version name is based on the name of the file and the version count, which is initialized to number and incremented each time output occurs. A true value for versions is the same as 1.

:tempo *number*
: The MIDI tempo of the file. Defaults to 60 bpm.

:divisions *integer*
: The number of MIDI divisions per quarter note. Defaults to 96.

:timesig (*numerator denominator*)
: The time signature for the file. Defaults to (4 4). The time signature list may contain up to four values: (numerator denominator clocks-per-quarter 32nds)

:channel-tuning {*false* | *true* | *1-16* | *list*}
: The default value of **`:channel-tuning`** is false, which means that floating point key numbers are rounded to the nearest MIDI integer keynum as they are sent to a MIDI file or port. If **`:channel-tuning`** is true then floating point key numbers *kkk.cc* will produce microtonal frequency resolution *cc* cents above the integer keynum *kkk*. In this case, channel assignments in MIDI events with floating point key numbers are ignored and the events are automatically routed to a channel determined by the system based on the value of **`:channel-tuning`**:

  ◊ If the value of **`:channel-tuning`** is boolean true then a floating point key number causes the current MIDI event to be preceded by a pitch bend and both messages are sent to the next available channel. By default, the entire range of channels 0-15 are cycled through in round-robin fashion to minimize the likelihood that a pitch bend value will affect a sounding note. To cycle through only a subrange of channels specify a list: (*true start end*) where *start* and *end* specify the starting and ending channel numbers in the rotation.

  ◊ If the value of **`:channel-tuning`** is an integer 1-16 then floating point key numbers will be quantized to that number of equal divisions per semitone. For example, **`:channel-tuning 2`** will quantize floating point key numbers to to the nearest quarter-tone. The maximum division of the semitone is 16, which produces a quantization step size of 6.25 cents, very close to the minimum perceivable change by humans. When channel tuning is expressed as an integer then that many channels are allocated and pretuned for microtonal quantization. By default this allocation starts at channel 0. To specify a different starting channel use a list: (*n start*), where *n* is the semitone division and *start* is the starting channel.

◊ If **`:channel-tuning`** is false then no microtonal output is produced and floating point key numbers are rounded to the closest integer key number.

Regardless of which channel tuning method is used, integer key numbers are never included in channel tuning output. By keeping a range of channel numbers free for normal tempered note-on and off messages both tempered and untempered sounds can be generated at the same time.

:pitch-bend-width *integer*

Declares the pitch bend width in semitones of the synthesizer. This value is used to calculate pitch bend values for microtonal frequency resolution. The default value is 2, which means that the synthesizer uses a maximum pitch bend range of one whole step up and down.

### Pattern Classes

The **`pattern`** class is the root class for defining musical patterns. Pattern-based composition is the topic of Chapter 20.

### Processes

A **`process`** is a special type of Lisp function that implements a procedural description of music. Processes are introduced in Chapter 13.

## Defining New Classes

The class hierarchy shown in Figure 1 may only be an approximation of the classes defined in Common Music. The reason for this is that Common Music provides the ability for composers to define their own classes of events and containers. Composers who work with *synthesis languages* will typically want to define customized sound events appropriate for the digital instruments they are using in the sound synthesis language. Common Music provides a macro function called **`defobject`** for defining new classes of objects and for specializing existing classes. The classes defined by **`defobject`** are automatically integrated into Common Music's score generation protocol. For more information on the **`defobject`** consult the Common Music Dictionary.

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file objects.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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H. Taube
06 May 2003

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# 12 *Etudes*, Op. 4: Score Generation

In *Etudes*, Op.4 we work thorough a series of examples that demonstrate how to generate scores using **`midi`** events, lists, and sequences.

## Op. 4, No. 1: The Current Working Directory.

Although is always possible to specify a score name that includes a directory pathn, all of the scores in this book specify just the name and extension of the score file. A file name without a directory component will be written to Lisp's current “working directory”. This directory can be accessed using the **`pwd`** function and specified using **`cd`**. Before generating any scores first make sure that Lisp's working directory is set to the directory where you would like to save score output to. You must have "write permission" in this directory.

[Function]
    (**pwd**)
[Function]
    (**cd** [*dir*])

The function **`pwd`** (print working directory) returns the string name of the current working directory. The function **`cd`** (change directory) changes the working directory . If the **`cd`** is called without its optional directory argument it sets the current working directory to whatever the operating system defines as the user's “home directory”. The directory specified to **`cd`** should be a string that ends with the directory character appropriate for the host operating system. For Linux, OSX and Windows use "/". For MacOS use ":" as the directory delimiter.

Interaction 12-1. Using the pwd and cd functions.

```
cm> (pwd)
"/home/hkt/"
cm> (cd "/tmp/")
"/tmp/"
cm> (pwd)
"/tmp/"
cm>
```

## Op. 4, No. 2: The events Function

The main score output processing function is called **`events`**. This function can be used to output events, sequences, processes and lists of these to a file, port, display or seq:

[Function]
    (**events** *objects score* [*ahead*] ...)

The **`events`** function is the main entry point for generating scores. *Objects* can a single object or a list of objects. *Score* can be a score or a filename. *Ahead* is an optional start time offset for *objects* in the score. If *ahead* is a single number it is applied to all the *objects*. If *ahead* is a list of numbers then each number is matched in left to right order with the corresponding object in *objects*. Following the optional offset may come any number of *keyword value* initializations suitable to *score*. **`events`** returns the name of the score as its value.

In our first example we output a single MIDI event to a MIDI score file. Before generating the file we use the **`pwd`** function to tell us what the current working directory is. If you do not have write permission in the directory that is returned by **`pwd`** then you can use **`cd`** to change to an appropriate output directory:

Interaction 12-2. Generating a MIDI event to a MIDI file.

```
cm> (define one (new midi :time 0
                   :keynum 60 :duration 2))

cm> (events one "myscore.midi")
"myscore.midi"
cm> (midi-file-print "myscore.midi")
File: myscore.midi
Format: 0
Tracks: 1
Division: 480

Track 0, length 20
       0 #<Tempo Change 1000 ms>
       0 #<Note-On 0 60 64>
     960 #<Note-Off 0 60 127>
       0 #<End of Track #(0)>
"myscore.midi"
cm>
```

## Op. 4, No. 3: Listening to MIDI Files

In this next exercise we listen to our score. Score files can be played from inside Common Music if there is an appropriate player defined for your operating system and audio/MIDI setup. The following MIDI file players can be used:

OSX Note:

[Function]

(**`osx-play-midi-file`** *file*)

plays a MIDI file using the QuickTime Player application.

Linux Note:

[Function]

(**`oss-play-midi-file`** *file*)

plays a MIDI file using the OSS utility /usr/bin/playmidi.

MacOS Note:

[Function]

(**`mac-play-midi-file`** *file*)

plays a MIDI file using the QuickTime Player application.

Windows Note:

[Function]

(**`win-play-midi-file`** *file*)

plays a MIDI file using the Windows Media Player application.

These functions can be called by hand or set as *output hooks* to automatically play score files after they are generated. In Interaction 12-3 we call the function by hand to listen to the score.

Interaction 12-3. Listening to a MIDI file on OSX.

```
cm> (osx-play-midi-file "myscore.midi")
"myscore.midi"
cm>
```

- myscore.midi

## Op. 4, No. 4: File Versions and Automatic Playback

It is possible to generate different versions of the same basic score and to automatically play each version after it has been generated by the **`events`** function. If *file versioning* is activated, the file name specified to the **`events`** function serves as a "base name" for creating numbered versions of the same score. For example, if the name of the score is "myscore.midi" and file versioning is activated, then successive calls to the **`events`** function will produce a sequence of files named "myscore-1.midi", "myscore-2.midi", "myscore-3.midi", and so on. Versioning is useful to ensure that no score will be accidentally destroyed by overwriting it with a newer version. It also enables the composer to listen to the different versions of a score as ideas are incrementally developed over several iterations of generation and playback. The file names that appear in the interaction examples of this book always reflect file versioning activated. The base names used for the scores may also vary according to chapter and section. This means that as you work through the examples and perform your own experiments, the file names that you see returned by the *events* function may not exactly match the file names printed in this book. The function that activates and deactivates file versioning for MIDI scores is called **`set-midi-file-versions!`**:

[Function]
    (**`set-midi-file-versions!`** *bool*)

Specifying true to **`set-midi-file-versions!`** enables file versioning, specifying false turns file versioning off.

It is possible to automatically play scores after they have been generated by setting an *output hook* that Common Music calls after the score is complete. The term “hook” describes a Lisp function that is associated with an automatic action.

[Function]
    (**`set-midi-output-hook!`** *fn*)

**`Set-midi-output-hook!`** establishes the function *fn* as a hook that called on MIDI files after they are generated by **`events`**. The hook function is passed the pathname (string) of the file and any additional hook arguments the user set in the file. To remove an existing output hook specify false in place of a function. In order to set a hook you must specify the actual function, not just its name. The special Lisp prefix **`#'`** “fetches” a function object given its name. In this next example we activate file versioning for MIDI files, and set an output hook for automatic score playback.

Interaction 12-4. Activating file versioning and setting an output hook for automatic playback.

```
cm> (set-midi-file-versions! true)

cm> (set-midi-output-hook! #'osx-play-midi-file)

cm>
```

Since these functions are used for an effect rather than for a value they do not return anything back to the Listener window for printing.

## Op. 4, No. 5: Generating score from Multiple Events

Multiple events can be generated to a score by specifying a list of objects to the **`events`** function. In this next example we use a loop to define a list of MIDI notes.

Example 12-1. Definition of a list of events.

```
(define up
  (loop for key from 60 to 72
        for beg from 0 by .1
        collect (new midi :time beg
                          :keynum key
                          :duration 1)))
```

Once the list is defined it can be passed to the **`events`** function to create a score.

Interaction 12-5. Generating a score from a list of events.

```
cm> (events up "myscore.midi")
"myscore-1.midi"
cm>
```

- myscore-1.midi

In Interaction 12-5 the variable *up* is defined to hold a list of **`midi`** events. The list is created using a loop that iterates two stepping variables. The first variable *key* is looped over every key number from 60 (Middle-C) to the octave above. The second variable *beg* starts at time 0 and is incremented by .1 seconds each time. Notice that an upper bound for this second iteration is not necessary since the first iteration variable includes one. For each value of *key* and *beg* the loop collects a new MIDI event. The overall effect is to generate a rapid, chromatic "strum" over one octave. This is the first example in the book that demonstrates a *procedure* for creating musical structure. Try redefining the *up* variable by a loop that iterates over different values for *beg* and *key*.

## Op. 4, No. 5: Mixing Objects

What if we would like to hear two simultaneous strums, one up and one down? A situation like this requires that two musical gestures overlap in the score. This is most naturally achieved by "mixing" output from the sequences. Before defining these objects we first generalize our loop by defining it as a function that creates either an ascending or descending chromatic strum at a specified rate, amplitude and duration:

Example 12-2. Create a list of midi events between two key numbers.

```
(define (strums key1 key2 rate dur amp)
  (let ((step (if (< key2 key1) -1 1))
        (diff (abs (- key1 key2))))
    (loop repeat (+ diff 1)
          for key from key1 by step
          for beg from 0 by rate
          collect (new midi :time beg
                            :duration dur
                            :amplitude amp
                            :keynum key))))
```

The **strums** function creates either an upward or downward strum between a starting *key1* and an ending *key2*. The direction of the iteration is controlled by the variable *step*. The value of *step* is determined by an **if** expression that tests to see if the ending key is lower that the starting key or not. If it is then -1 is returned and the iteration proceeds downward from *key1*. If *key2* is greater than *key1* than the iteration ascends from *key1* by 1. The variable *diff* holds the number of chromatic steps between the two keys. The **abs** function is used get the absolute value of the difference since the difference would otherwise be negative if *key2* were less than *key1*. The loop then repeats *diff+1* times so that the last note will actually contain the value of *key2*. The rest of the loop is similar to the first example. In this next example we look at the output from a call to **strums** to see the list it computes. **pprint** displays the list in a "pretty format" that is easy to read in the Listener window.)

Interaction 12-6. Generating strum data.

```
cm> (pprint (strums 60 61 .1 1 .2))
(#i(midi time 0 keynum 60 duration 1 amplitude 0.2 channel 0)
 #i(midi time 0.1 keynum 61 duration 1 amplitude 0.2 channel 0))
cm>
```

We next create two sequences to represent two different strums and then mix them together to form the score.

Interaction 12-7. Creating two strum sequences.

```
cm> (new seq :name 'ups
         :subobjects (strums 48 60 .1 1 .4))
#<seq "ups">
cm> (new seq :name 'downs
         :subobjects (strums 84 72 .1 1 .2))
#<seq "downs">
cm> (events (list #&ups #&downs) "myscore.midi")
"myscore-2.midi"
cm>
```

- myscore-2.midi

The **list** function creates a list containing the sequences named "up" and "down" that is passed to the **events** function. The special prefix **#&** “fetches” a container from its name.

## Op. 4, No. 6: Start Time Offsets

This final example demonstrates the effect that start time offsets have on **events** generated to a score. If offsets are specified they should appear directly after the score argument in the **events** call. In the first interaction of Interaction 12-8 the onset of both strums is delayed by 5 seconds in the score. In the second interaction each strum receives a different offset, 0 for the up strum and .5 for the down. The last example shows multiple instances of the strums being scheduled at different times.

Interaction 12-8. The effect of start time offsets.

```
cm> (events (list #&ups #&downs) "myscore.midi" 5)
"myscore-3.midi"
cm> (events (list #&ups #&downs) "myscore.midi" '(0 .5))
"myscore-4.midi"
cm> (events (list #&ups #&downs #&ups #&downs )
            "myscore.midi"
            '(0 .5 1 1.5))
"myscore-5.midi"
```

```
cm>
```

- myscore-3.midi
- myscore-4.midi
- myscore-5.midi

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file scores.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 13 Algorithms and Processes

An algorithm must be seen to be believed, and the best way to learn about what an algorithm is all about is to try it.

— Donald Knuth, The Art of Computer Programming

In this chapter we begin to explore *algorithmic composition*, the procedural description of musical scores. In this style of composition the composer writes a program to compute music. The program introduces a level of abstraction between the composer and the score that allows the composer to explore new techniques for music composition that would otherwise be difficult or impossible to achieve. The use of algorithms and the level of indirection they represent also raise some issues concerning the relationship of the composer to the generated score:

- What is the relationship between a composer and a computed score? If the composer cannot predict what notes will be contained in the score should he or she claim authorship of the composition?
- When an algorithm creates a score is it really composing, does it mimic composition or neither? If a person cannot tell the difference by listening to the score or by studying it, does it matter?
- Should a listener try to appreciate algorithmic constructions aesthetically apart from their sound?
- What happens when a computer program creates the best piece on the concert? Should the audience clap?

Before turning our attention to some specifics of algorithmic composition we should first define what an algorithm is and how it relates to the generation of musical material. In his monumental opus The Art of Computer Programming, Donald Knuth defines an *algorithm* as a set of steps, or rules, with five basic properties:

1. An algorithm must start and stop. The rules an algorithm applies must also conclude in a reasonable amount of time. What “reasonable” is depends on the nature of the algorithm, but in no case can an algorithm take an infinite amount of time to complete its task. Knuth calls this property the *finiteness* of an algorithm.
2. The actions that an algorithm performs cannot be open to multiple interpretations; each step must be precise and unambiguous. Knuth terms this quality *definiteness*. An algorithm cannot iterate a "bunch" of times. The number of times must be precisely expressed, for example 2, 1000000, or a randomly chosen number.
3. An algorithm starts its computation from an initial state. This state may be expressed as input values given to the algorithm before it starts.
4. An algorithm must produces a result with a specific relation to the inputs.
5. The steps an algorithm takes must be sufficiently simple that they could be expressed “on paper”; these steps must make sense for the quantities used. Knuth terms this property *effectiveness*.

These points can be illustrated using the **`strum`** procedure presented in Chapter 10 and shown again here in Example 13-1 under a slightly different name.

Example 13-1. The strum1 definition.

```
(define (strum1 key1 key2 rate dur amp)
  (let ((step (if (< key2 key1) -1 1))
        (diff (abs (- key2 key1))))

    (loop repeat (+ diff 1)
          for key from key1 by step
          for beg from 0 by rate
```

```
      collect (new midi :time beg
                        :duration dur
                        :amplitude amp
                        :keynum key))))
```

The **strum1** function adheres to Knuth's five characteristics as follows:

1. Strum1 exhibits finiteness by iterating a specific number of times as determined by its starting and ending key numbers. The loop's repeat clause expresses finiteness quite succinctly.
2. Strum1 is definite because it consists of a series of Lisp expressions in which every quantity and relation is well defined.
3. Strum1 has five input parameters that determine the initial state of the algorithm.
4. Strum1 produces a result, the list of musical events it returns to the caller.
5. Strum1 is effective because it is able to compute its result from its inputs and rules.

As we can see in this example, functions can provide a natural mechanism for expressing algorithms. When an algorithm is implemented as a Lisp function it is no longer just a series of steps or rules — it is a true Lisp object that be manipulated and processed like all other constructs in the language. This means that it is possible to test and judge algorithms as entities in their own right. The criteria for judging an algorithm might include its effectiveness, speed, complexity, or even elegance of design.

## Musical Processes

The algorithms we will explore in this book are implemented as Lisp functions. One special type of function, called a *process*, will be used more than any other tool to express procedural composition. A process is a Lisp function that computes musical structure — sound events, sequences, even other processes — *dynamically*, while the **events** function is generating a score. A process represents the evolution of a compositional idea over time. Processes execute in such a way that many different processes may run in parallel, or at the same time. Each process can be thought of as an iteration of a function over a single time line in the score that may coexist with other time lines. These parallel processes are managed by a surrounding environment called a *scheduler*.

### The Scheduler

A scheduler is like a conductor, its job is to insure that the musical objects under its control are evaluated in a systematic order and at their correct times in a score. Scheduling is initiated by the **events** function when it is called with a list of objects to process and a score to create. The scheduler generates a time ordered series of events for the score by dynamically sorting the objects under its control in a *priority queue*. The priority of each object is determined by the value of the **time** slot contained in the object. Objects toward the front of the queue will occur earlier in the score than objects further back. If two objects have the exact same score time then the object that was placed in the queue first has more priority than the other object at that same time. The scheduler creates a score by *handling*, or processing, the objects in its scheduling queue. The scheduler "handles" an object by first removing it from the head of the queue, and then processing it in some manner specific to the type of object and score being handled. Handling one object may result in new objects being inserted into the scheduling queue, or in the same object being reinserted into the queue with a new priority. The scheduler continues to handle the objects in its queue until no more objects remain to be processed. At that point score generation is complete and the scheduler stops.

#### Event Handling

For a musical event the handling process results in the event being sent to the score that is currently being generated. The effect of sending depends both on the type of event and the type of score being generated. For example, a MIDI event that is sent to a MIDI score results in a number of MIDI byte messages being written to a MIDI file on the disk. A MIDI event that is sent to a seq results in the event itself being inserted into the list of subobjects contained in the seq.

### Sequence Handling

The scheduler handles a seq object by descending into its subobjects and adding each of these objects into the scheduling queue relative to the priority time of the seq object itself.

### Process Handling

The scheduler evaluates a process function by calling it to run its program. While there is no restriction on what a process function does, typical actions include:

- creating musical events and sending them to the score.
- creating new structures (events, sequences and processes) and adding them to the priority queue.
- determining if the process should run again and, if so, determining what its next priority time will be.

## Defining Musical Processes

A process function is created using the **`process`** macro function. A **`process`** definition looks almost identical to a **`loop`** definition except that the action clauses in a process implement scheduling and score action tasks rather than accumulating values. The **`loop`** function returns values to the caller, the process function returns a function that will be run inside the scheduler.

We will define processes inside of Lisp functions and then call Lisp functions to create process. We do this for two important reasons:

- A function encapsulates the process definition in a named, reusable abstraction. The function can be called multiple times in a program without having to type the process definition again. Each time the function is called it will return a new instance of the process it contains as its value.
- If the function has arguments, different versions of the same basic process description can be created simply by calling the function with different input values. This provides a very simple “template” mechanism the produce variations without any extra work.

In all cases we will avoid storing processes in global variables. A process only “lives” inside the scheduler, once the scheduler has finished all of the processes inside it are or no more use. If you store a process in a global variable you will have to remember to redefine that variable to a new process function each time you use it. This is much more work than simply calling a function. Before describing the specifics of the **`process`** macro we will first look at two examples of process descriptions.

## Converting strum1 Into a Process

We start with an example that simply converts the **`strum1`** definition in Example 13-1 to return a process object instead of a list:

Example 13-2. Converting the strum algorithm into a process description.

```
(define (strum2 key1 key2 rate dur amp)
  (let ((step (if (< key2 key1) -1 1))
        (diff (abs (- key2 key1))))
```

```
    (process repeat (+ diff 1)
             for key from key1 by step
             for beg from 0 by rate
             output (new midi :time beg
                          :duration dur
                          :amplitude amp
                          :keynum key))))
```

The definition of **strum2** is almost the same as **strum1** in Example 13-1 except for three very simple modifications:

1. The name of the function has been changed from **strum1** to **strum2**.
2. The **process** function appears in place of the **loop** function.
3. An **output** action clause appears in place of a **collect** action clause.

We can now use the definition of **strum2** to generate a score from a process:

Interaction 13-1. Using a strum2 process.

```
cm> (events (strum2 48 60 .1 1 .4) "strum.midi")
"strum-1.midi"
cm>
```

- strum-1.midi

## Using now and wait

In Example 13-3 **strum2** is rewritten to better reflect the dynamic nature of processes:

Example 13-3. Using the now and wait functions.

```
(define (strum3 key1 key2 rate dur amp)
  (let ((step (if (< key2 key1) -1 1))
        (diff (abs (- key2 key1))))
    (process repeat (+ diff 1)
             for key from key1 by step
             for beg = (now)
             output (new midi :time beg
                          :duration dur
                          :amplitude amp
                          :keynum key)
             wait rate)))
```

The **strum3** process differs from **strum2** only in the way that time is handled. In **strum2** the start time of each event is determined by *beg*, which stepped from 0 by the *rate* specified to the function. This caused the time of each generated event to change but the time of the process itself remained unchanged. **strum3** represents an alternate strategy in which the start time of the process is incremented and the MIDI events are always output at the current time of the process, whatever that may be. This allows the process to “evolve” over time inside the scheduler along with the data it generates.

In the new version of **strum3** the **now** function is used to return the current priority time of the scheduler and the *beg* variable is set to this value. *beg* is the time stamp for MIDI objects sent to the score. The **wait** action causes the scheduler to reinsert the current process back into the scheduling queue *rate* seconds later than its current time.

Interaction 13-2. Listening to strum3.

```
cm> (events (strum3 84 72 .1 1 .4) "strum.midi")
"strum-2.midi"
cm> (events (list (strum3 48 60 .1 1 .4)
                  (strum3 84 72 .1 1 .2))
            "strum.midi")
"strum-3.midi"
cm>
```

- strum-2.midi
- strum-3.midi

Note that in the definition of the different strum processes we did not need to create lists of events or define seq objects to hold separate musical time lines. Processes can produce the same effects dynamically, while the score is being generated, without the overhead of creating sequences and lists ahead of time. This also means that the actions taken by processes can change dynamically, on the fly, in response to dynamic conditions in the score.

## The process Macro

The **`process`** macro supports the description of iterative musical processes using the same basic syntax as the **`loop`** macro introduced in Chapter 8. While both process and loop perform iteration, a process and a loop are not equivalent constructs. A loop executes immediately and returns values back to the caller. The process macro, on the other hand, does not iterate — it defines a special type of function that is *scheduled* to iterate and generate music. The **`loop`** macro returns values to the caller, the **`process`** macro returns a function to execute inside the scheduler.

A **`process`** iteration is defined as a series of clauses. Like loop, each clause in a process definition begins with a clause operator. The **`process`** macro defines a set of action clauses that are specific to music composition; these action clauses are documented below. The other types of clauses — iteration, conditional, and finalization — are identical to those provided by **`loop`** and are documented in Chapter 8.

### Process Action Clauses

Process action clauses implement score and scheduler tasks. Unlike **`loop`** actions, process actions do not return values.

[Clause]
    **`output`** *event* `[`*score*`]`
[Clause]
    **`wait`** *delta*
[Clause]
    **`sprout`** *object* `[at` *ahead*`]`
[Clause]
    **`do`** *expr ...*
[Clause]
    **`set`** *var* `=` *value*
[Clause]
    **`each`** *var* `{in` *...* `|` `from` *...*`}` `[as` *...*`]` *action*

The **output** clause sends *event* to *score*, which defaults to the score specified to the **events** function. **Wait** sets the next priority run time of the process to *delta* seconds later. The **sprout** clause adds *object* to scheduling queue. One *object* or a list of objects may be specified. An optional **at** specifies a future *ahead* time. If **at** is not supplied then *object* is inserted into the scheduling queue at the current process time. The **do** clause evaluates one or more Lisp expressions and is equivalent to the identical clause in **loop**. The **set** clause sets a variable *var* to *value* but does not actually define *var* inside the process. Use **set** to set variables that are established outside the lexical scope of the process but are set inside it.

The **each** clause defines an iteration over the *action* clause. It supports the same stepping clauses that the **for** iteration clause supports. The difference between **for** and **each** is that the former defines a process iteration while **each** defines an iterative action inside the process.

## Process Examples

We now look at several examples of process definitions that demonstrate different process action clauses. The first few examples require several of the chance and 12-tone functions we implemented in Chapter 7 and Chapter 8. These are shown again here in Example 13-4.

Example 13-4. Helper Functions from Chapter 6 and 8.

```
(define (chance? prob)
  (< (random 1.0) prob))

(define (keynum->pc k)
  (mod k 12))

(define (retrograde-row row)
  (reverse row))

(define (transpose-row row to)
  (loop for pc in row
        collect (keynum->pc (+ pc to))))

(define (invert-row row)
  (loop for pc in row
        collect (keynum->pc (- 12 pc))))
```

### A Simple Row Process

In the first example we implement a simple process to play twelve-tone rows.

Example 13-5. Basic row playing.

```
(define row1 '(0 1 6 7 10 11 5 4 3 9 2 8))
(define row2 (invert-row row1))
(define row3 (retrograde-row row1))
(define row4 (retrograde-row row2))

(define (ttone1 reps row key beat amp)
  (process with len = (length row)
           repeat reps
           for i from 0
           for p = (mod i len)
           for pc = (list-ref row p) ; i mod 12
           for k = (+ key pc)
```

```
          output (new midi :time (now)
                      duration (* beat 2)
                      :keynum k
                      :amplitude amp)
          wait beat))
```

The **`ttone1`** process plays *reps* number of events given a *row* of pitch classes, a *key* number offset, a rhythmic *beat* and an *amp* loudness factor. The stepping variable *i* is a counter that increments by 1 each time the process runs. The variable *pc* is set to the current pitch class at position *p* in *row*. Recall that **`list-ref`** returns an element from a list at a specified index and that the first element in a list is always at index 0 and the last element is at index *len-1*. Our index into *row* is determined by taking the mod 12 value of *i*. Although this length is always 12 in our examples we avoid specifying the value 12 directly so that it is possible to listen to pitch class lists with more or fewer than twelve elements. The **`mod`** calculation insures that our index *p* will always cycle over valid list positions from 0 to 11 as *i* continually increases by 1. The key number *k* for the current event is calculated by adding the value of the pitch class *pc* to the *key* offset that was input into the function. The duration of each event is calculated to be *2*beat* so that a legato effect is produced no matter what value of *beat* is specified to **`ttone1`**. The rest of the example is similar to earlier processes definitions presented in this chapter.

Interaction 13-3. Listening to ttone1

```
cm> (events (ttone1 36 row1 60 .2 .5)
            "ttone.midi")
"ttone-1.midi"
cm>
```

- ttone-1.midi

## Variation 1

In this next example we make simple variation of **`ttone1`** that plays a “pointillistic” version of a row.

Example 13-6. A pointillistic row.

```
(define (ttone2 len row key beat amp)
  (process repeat len
          for i from 0
          for pc = (list-ref row (mod i 12))
          for n = (if (chance? .5)
                      (+ key 12)
                      (- key 12))
          for k = (+ n pc)
          output (new midi :time (now)
                      :duration (* beat 2)
                      :keynum k
                      :amplitude amp)
          wait beat))
```

The main difference between this example and the preceding one is that **`ttone1`** determines a key number *k* by randomly choosing between values one octave up or down from the *key* input into the function. This has the effect of slightly randomizing the register of the row so that it does not sound quite as mechanical as the row repeats over the course of the process. The transposition value *n* is chosen by calling the **`chance?`** function we defined in Chapter 7. The expression **`(if (chance? .5) (+ key 12) (- key 12))`** says that 50% of the time *n* will be *key+12* and 50% of the time it will be *key-12*.

Interaction 13-4. Listening to ttone1

```
cm> (events (ttone2 36 row1 60 .2 .5)
            "ttone.midi")
"ttone-2.midi"
cm>
```

- ttone-2.midi

## Variation 2

The next variation improves on the preceding one by implementing different wait times and controlling the amount of random pointillism that is applied to the row.

Example 13-7. A pointillistic row.

```
(define (ttone3 len row key beat amp)
  (process repeat len
           for i from 0
           for pc = (list-ref row (mod i 12))
           for w = (* beat (random 4))
           for n = (if (= w 0)
                     (if (chance? .5)
                       (+ key 12)
                       (- key 12))
                     key)
           for k = (+ n pc)
           output (new midi :time (now)
                       :duration (* beat 2)
                       :keynum k
                       :amplitude amp)
           wait w))
```

In Example 13-7 the *w* stepping variable is set to a random multiple of *beat*. Recall that the expression **`(random 3)`** generates random integers from 0 less than 4. This means the wait value *w* will be either 0 (when random returns 0), *beat* (when random returns 1), *2*beat* or *3*beat*. A zero wait value causes the next *run time* of the process to be the same as its current time in the score. This means that the events generated over separate iterations will be simultaneous in the score. The last improvement that **`ttone3`** makes is to transpose the key number only in the case of simultaneous notes. This has the effect of preserving some “registeral integrity” of the row's melodic contour while still providing for some variety.

Interaction 13-5. Listening to ttone3

```
cm> (events (list (ttone3 36 row1 40 .2 .5)
                  (ttone3 36 row1 60 .2 .5))
            "ttone.midi"
            '(0 4))
"ttone-3.midi"
cm>
```

- ttone-3.midi

## Variation 3

In our final twelve-tone example we modify the process definition to improve sound characteristics when simultaneous instances of the process are running.

Example 13-8. Defining sections and instrumentation.

```
(define (ttone4 dur row key beat amp chan)
  (process while (< (now) dur)
           for i from 0
           for pc = (list-ref row (mod i 12))
           for r = (* beat (random 4))
           for n = (if (= r 0)
                     (if (chance? .5)
                       (+ key 12)
                       (- key 12))
                     key)
           for k = (+ n pc)
           output (new midi :time (now)
                       :duration (* beat 2)
                       :keynum k
                       :amplitude amp
                       :channel chan)
           wait r))
```

**`ttone4`** makes two small modifications that greatly improve the musicality of the output. The first modification is the addition of a new *chan* parameter to the function to allow each process to output to a MIDI channel specified by the composer. MIDI channels controls which instrument in the synthesizer receives the notes. Remember that indexing always starts from 0 in Lisp so MIDI channels in Common Music are numbered from 0 to 15. This allows channel values to be calculated without always having to add or subtract 1.

The second modification to the process changes the way in which the algorithm controls iteration. In all of the preceding examples the processes iterate an exact number of times. In this version, the process iterates for a specified *duration* of time, regardless of how may iterations the process executes. The conditional clause

```
      while (< (now) dur)
```

says to run the process as long as the current time of the process is less than the duration specified to the function. As soon as this condition is false the scheduler automatically stops the process.

We now listen to three contrapuntal voices of the same basic process definition, where each version has a different row form, register, instrument, start time and duration in the score. All three processes stop generating events at the same time in the score. Before playing the example be sure to select different MIDI instruments on the first three channels of your synthesizer. Listen to several versions of the example to observe the large scale effects that the random selection has on each process.

Interaction 13-6. Playing a little 12-tone etude.

```
cm> (events (list (ttone4 16 row1 40 .2 .5 0)
                  (ttone4 12 row2 60 .2 .5 1)
                  (ttone4 8 row3 80 .2 .5 2))
            "ttone.midi"
            '(0 4 8))
"ttone-4.midi"
cm>
```

- ttone-4.midi

## Ghosts

As a final example of algorithmic process definition we look at a more involved definition called **`ghosts`**, written by Tobias Kunze. In addition to its interesting musical properties **`ghosts`** also provides examples of most of the action clauses supported by the **`process`** macro. The definition of **`ghosts`** requires several “helper functions” that were introduced in Chapter 7. The definitions of these auxiliary functions are shown again here in Example 13-9.

Example 13-9. Helper functions for ghosts.

```
(define (pick-list lst)
  (let* ((len (length lst))
         (pos (random len)))
    (list-ref lst pos)))

(define (pick-range low high)
  (let ((rng (- high low)))
    (if (> rng 0)
      (+ low (random rng))
      0)))

(define (bpm->seconds bpm)
  (/ 60.0 bpm))

(define (rhythm->seconds rhy tempo)
  (* rhy 4.0 (bpm->seconds tempo)))
```

The **`ghosts`** process gets its name from the fact that a short, randomly generated melody serves as the basis for computing other bits of dependent **`ghost`** structure that echo the melody in different ways at larger temporal levels in the piece. **`ghosts`** generates the following gestures, each of which depends on certain tests applied to the characteristics of the main melody:

- High, temporally stretched versions of melody tones that occur in the upper range of the main melody.
- Low, percussive thumps that accompany low main melody tones that have a certain duration.
- Distant strums that accompany the high stretched melody at even larger time scales.

Before looking at the main process we first define the functions that create the dependent gestures used in **`ghosts`**.

Example 13-10. Defining gestures for ghosts.

```
(define (hitone knum at)
  ;; create a long tone two octave above knum
  (new midi :time at
       :keynum (+ knum 24)
       :duration at
       :amplitude .5))

(define (thump knum at)
  ;; create short percussive events below knum
  (list (new midi :time at
              :keynum (- knum 18)
              :duration .05 :amplitude .4)
        (new midi :time at
             :keynum (- knum 23)
             :duration .05 :amplitude .4)))

(define (riff knum rhy)
  ;; generate upward strum of notes
  (let ((rate (rhythm->seconds (/ rhy 4) 60)))
    (process repeat 5
             for k from (+ 39 (mod knum 13)) by 13
             output
             (new midi :time (now) :keynum k
                  :amplitude .3 :duration 10)
             wait rate)))
```

The **`hitone`** function creates a single **`midi`** event two octaves (24 semitones) above the *knum* specified to the function. *At* serves as both the start time of the new **`midi`** event as well as its duration. The effect of this is to create longer and longer “high tones” as the **`ghost`** process runs.

The **`thump`** function returns a list of two **`midi`** notes, each with a very short duration. *knum* is the current main melody note and *at* is the time at which the thump events are to occur in the score. The key number of the first thump event is always an octave and a tritone lower than the main melody note; the second event is a perfect fourth above that. Since these transpositions are fixed relationships the thump events will follow the general melodic contour of the main melody notes to which they are attached.

The **`riff`** function defines a process to play 5 rapid events in an upward arc. *knum* is the current main melody note and *rhy* is its metric rhythm value. **`riff`** converts this metric rhythm into a wait value that is four times faster then the main melody rhythm. The first event that **`riff`** plays is a transposition of the main melody key number shifted to lie somewhere within the octave above key number 39 (E-flat in the second octave). The remaining four events that the process creates each have key number a minor-9th above the previous event.

We can listen to the gestures of **`thump`** and **`riff`** in isolation before proceeding to the main process definition:

Interaction 13-7. A sample thump and strum.

```
cm> (events (hitone 60 0) "thump.midi")
"thump-1.midi"
cm> (events (thump 60 0) "thump.midi")
"thump-2.midi"
cm> (events (riff 60 1/8) "riff.midi")
"riff-1.midi"
cm>
```

- thump-1.midi
- thump-2.midi
- riff-1.midi

The **`ghosts`** process generates a main melody in the middle register of the virtual MIDI keyboard and then *sprouts* dependent structure above and below the main melody based on tests it applies to the main material. Recall that the **`sprout`** action clause inserts objects into the scheduling queue at a future time. By using **`sprout`** instead of **`output`** the dependent structures that **`ghosts`** generates reflect current conditions of the process that are actually heard in the score at a much later point in time.

Example 13-11. The ghosts process.

```
(define (ghosts)
  (process repeat 12
           for here = (now)
           for ahead = (* (+ here .5) 2)
           for main = (pick-range 53 77)
           for high? = (>= main 65)
           for amp = (if high? .6 .4)
           for rhy = (pick-list '(1/16 1/8 3/16))
           ;; output main melody
           output (new midi :time here
                       :keynum main
                       :duration (rhythm->seconds rhy 60 )
                       :amplitude amp)
           when high?
           sprout (hitone main ahead)
           and sprout (riff main rhy) at (* ahead 2)
           when (= rhy 3/16)
           sprout (thump main (+ here .5))
           wait (rhythm->seconds rhy 60 )))
```

The **`here`** variable holds the current score time returned by **`now`**. *Ahead* is a future time stamp that is twice the value of one half second later than the current time in the score. This simply means that as *here* increases each time the process runs, *ahead* is roughly twice as far into the future. *Main* is the main key number of the melody and is randomly set to any key number within a two octave range from 53 (F below Middle C) to 77 two octaves above it. The *high?* variable is set to either true or false based on a test which the process applies to the *main* key number. Recall that arithmetic relations in Lisp are functions that return either true or false. This means that the expression **`(>= main 65)`** returns boolean true if the *main* key number is greater than or equal to 65 otherwise it returns false. Since *high?* is set to the value of this expression the *high?* variable is true if the main key number was selected from the upper octave in the key number range, otherwise *high?* is false. The amplitude value *amp* of the main melodic tone is dependent on whether or not the *main* key number is high or not. If *high?* is true then the main melody is played with at an amplitude of .6 otherwise it is played at the softer value .4. The last stepping clause sets the rhythm for the main melody *rhy* to a random choice of either a sixteenth, eight or dotted-eighth value.

The first action that **`ghosts`** performs is to output a main melody **`midi`** event at the current time in the score. The next clause in the process definition is a *conditional* clause that executes two sprout actions only if the *high?* variable is true. The conditional expression can be reduced to show its key clausal features:

when *high?* sprout *<hightone>* and sprout *<riff>*

The **`when`** clause performs a test and evaluates one or more dependent action clauses only if the test is true. The **`and`** conjunction joins action clauses together that depend on a single test. The conditional clause in **`ghosts`** has the effect of generating a high melodic tone 24 semitones above the main tone and a riff gesture only if the current *main* tone lies within the upper octave of its register. The next clause in the process is another conditional clause but in this case the test applies a current rhythmic value and there is only one dependent action clause.

when (= *high* 3/16) sprout *<thump>*

The effect of this clause is to only create a thump gesture if the current value of *rhy* is 3/16, i.e. a dotted-eighth value. The last clause in **`ghosts`** is a **`wait`** action clause that increments the priority time of the process by the current rhythmic value of *rhy* converted to seconds. Since there is no **`and`** conjunction preceding the wait clause it is independent of the preceding conditional and so it evaluated every time the process runs.

We are now ready to listen to **`ghosts`**. Generate the score a number of different times to hear how the non-random dependent gestures track the randomly generated main melody.

Interaction 13-8. Listening to **`ghosts`**.

```
cm> (events (ghosts ) "ghosts.midi")
"ghosts-1.midi"
cm> (events (ghosts ) "ghosts.midi")
"ghosts-2.midi"
cm>
```

- ghosts-1.midi
- ghosts-2.midi

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file processes.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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06 May 2003

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# 14 *Etudes*, Op. 5: Steve Reich's *Piano Phase*

In Etudes Op. 5 we model some of the basic algorithmic features of Steve Reich's composition *Piano Phase*. The purpose of this Etude is not to recreate a complete and faithful rendition of Reich's wonderful composition, but rather, to try to understand some of the issues involved with his notion of *phasing*, and to see how these issues might be represented in the metalevel. The implementation also serves as a mechanism for experimenting with different phasing controls and provides a basis for modifications and improvements the reader may care to implement.

## The Composition

*Piano Phase* is a composition for two pianists that lasts approximately twenty minutes. (The reprinted score and directions to *Piano Phase* can be found on page 554 of Tonal Harmony by Kosta and Payne.) The score and directions are quite simple. Both pianists play a single group of notes, which we will call the *trope*, over and over again for the entire length of the composition. The piece is laid out in a series of sections, the exact length of each section — measured in repetitions of the trope — is left up to the performers within limits that Reich sets out in the score.

*Piano Phase* begins with only the first pianist playing the trope at a constant tempo. The first performer will continue playing in this manner for the entire composition. The second pianist waits for several more repetitions of the trope and then joins in at the unison. After remaining "in phase" for several repetitions of the trope the second pianist increases the tempo over the course of several repetitions of the trope, until their part is one note ahead of the first performer. At this point the second pianist resumes the constant tempo again so that the two performers are in lock step but now one note out of phase with each other. This process continues until the second performer has moved ahead through the entire trope. In the final section of the composition both performers are once again playing the trope in unison at the identical constant rate.

## The Implementation

In order to make the code as understandable as possible, our implementation of *Piano Phase* will provide the basic looping and phasing features without implementing randomly selected section lengths or a slowly increasing tempo. These improvements will be left to the reader to design once patterns and linear interpolation have been covered in later chapters. Out first job will be to implement the musical process that represents Piano 1. Since this performer plays the trope without any change for the length of the composition, the basic process definition is quite simple:

### Opus 5, No. 1: Piano 1

Example 14-1. Helper functions for Piano Phase.

```
(define pp-trope
  (keynum '(e4 fs b cs5 d fs4 e cs5 b4 fs d5 cs5)))

(define pp-pulse 1/24)

(define pp-tempo 72)

(define (bpm->seconds bpm)
  (/ 60.0 bpm))

(define (rhythm->seconds rhy tempo)
  (* rhy 4.0 (bpm->seconds tempo)))

(define (piano1 trope amp chan)
  (let* ((tlen (length trope))
         (cycs tlen)
         (rate (rhythm->seconds pp-pulse pp-tempo)))

    (process repeat (* tlen cycs)
             for i from 0
             for x = (mod i tlen)
             for k = (list-ref trope x)
             output (new midi :time (now)
                         :keynum k
                         :duration (* rate 1.5)
                         :amplitude amp
                         :channel chan)
             wait rate)))
```

The **`piano1`** process repeats a *trope* of key numbers for as many times as there are key numbers in the trope. The total number of events that the process plays is equal to length of the trope *tlen* times *cycs*, the number of times the trope is repeated. The variable *x* is the index into *trope* of the current key number to play. *x* is set to *i* mod *tlen* so that *x* continually cycles from zero to *tlen*-1 as *i* increases by one each time the process runs. The *rate* at which the process plays is determined by multiplying the global pulse value *pp-pulse* by the global tempo factor *pp-tempo*.

Interaction 14-1. Listening to 12 repetitions of pp-trope.

```
cm> (events (piano1 pp-trope .5 0) "reich.midi")
"reich-1.midi"
cm>
```

- reich-1.midi

## Opus 5, No. 2: Piano 2

The function that represents the second performer is responsible for implementing the regular pattern of phasing that occurs throughout the piece. The phasing pattern consists of two parts. The first part is characterized by a constant rate of events at the identical speed as **`piano1`**. In the second part of the phasing the trope is played slightly faster than the first performer until by the end of the part **`piano2`** is one note ahead of **`piano1`**. At this point the phasing pattern starts all over again. The phasing pattern can be represented using a *tempo curve*, a list of tempo coefficients that will be applied to each event to produce a dynamically changing rate for the process. The coefficients in the first half of the tempo curve will consists of 1's. The coefficients in the second half of the tempo curve will consist of values slightly less than 1. The coefficients will be less than 1 because they will scale the *rate* value that the process waits for — the smaller the coefficient the shorter time the process waits between events.

The phasing coefficients can be represented as a ratio *p/n* where *p* is the time the phasing takes (counted in pulses) and *n* is the number of events to play in that time. For example, a *p/n* ratio of 16/16 would mean to play 16 notes in the time of 16 pulses and a *p/n* of 16/17 would mean to play 17 notes in the time of 16 pulses. If we assume that the each half of the phasing equals one repetition of the trope, then the *p/n* ratio for our version of Piano Phase would be 12/13, where the *p* value of 12 is the length of Reich's trope and *n* is one more than that value. That extra *n* value is the note that the performer moves ahead by. We can now implement a function that creates a tempo curve:

Example 14-2. The pp-tempo-curve definition.

```
(define (pp-tempo-curve tlen stay move)
  (let* ((p (* tlen move))
         (n (+ 1 (* tlen move)))
         (l1 (loop repeat (* tlen stay)
                   collect 1))
         (l2 (loop repeat (+ 1 (* tlen move))
                   collect (/ p n))))
    (append l1 l2)))
```

**`pp-tempo-curve`** returns a list of tempo coefficients appropriate for scaling the rate of events that the **`piano2`** process will produce. *len* is the length of the trope, which is 12 in the case of the trope used in *Piano Phase*. *Stay* is the number of repetitions of the trope that **`piano2`** is synchronized with **`piano1`**. *Move* is the number of repetitions of the trope during which **`piano2`** plays faster than **`piano1`**. The *p* and *n* values are calculated from these repetition values times the length of the trope. The *l1* variable holds the list of tempo values for the first half of the phasing. The coefficients for the first half are all 1. The *l2* variable holds the list of tempo values for the second half of the phasing. The coefficients in this list are the *p/n* ratio that produce the faster tempo. The second half of the list contains 1 more element than the first half. The tempo curve list is created by appending the two halves of the curve together to form one large list that is returned as the value of the function. We test the tempo curve function with small input values to see what it produces.

Interaction 14-2. Creating a tempo curve list.

```
cm> (pp-tempo-curve 3 1 1)
(1 1 1 0.75 0.75 0.75 0.75)
cm>
```

The first half of the tempo curve contains three 1's which add up to 3 beats. The second half of the tempo curve list contains four .75 values which also add up to 3 beats. The total duration of the phasing sequence is 6 beats, so this tempo curve process would play seven events in the same time that a constant process at tempo 1 played 6 events with both processes synchronized for the first three events. Given this tempo curve function we are now ready to implement **`piano2`**.

Example 14-3. The piano2 process.

```
(define (piano2 trope stay move amp chan)
  (let* ((tlen (length trope))
         (cycs tlen)
         (coda (* stay tlen))  ; ends with unison segment
         (curve (pp-tempo-curve tlen stay move))
         (clen (length curve))
         (rate (rhythm->seconds pp-pulse pp-tempo)))
    (process repeat (+ (* clen cycs) coda)
             for i from 0
             for k = (list-ref trope (mod i tlen))
             for c = (list-ref curve (mod i clen))
             output (new midi :time (now)
                          :keynum k
                          :duration (* rate 1.5)
                          :amplitude amp
                          :channel chan)
             wait (* rate c))))
```

*Trope* is the list of key numbers to play. *Stay* and *move* are the two parts to the phasing pattern — *stay* is the number of repetitions of *trope* during which **`piano2`** remains at the constant tempo, *move* is the number of repetitions of *trope* during which the tempo is increased. *tlen* and *cycs* are exactly the same as in **`piano1`**. *Coda* is a number of events to play at the very end of the piece, after *cycs* number of pattern cycles has occured and the entire process has moved through all the notes of the trope. The extra *coda* amount allows both pianos to end playing the constant portion of the pattern in unison, just as the piece begins. *Curve* is the tempo curve and *clen* is the length of it, which is also the total number of notes in the phasing pattern. The body of the process definition is very much like **`piano1`**, except that, in addition to getting the current note from *trope* **`piano2`** also gets the current tempo coefficient *c* from the tempo curve. This coefficient is then used to scale the global rate value in the wait expression.

Interaction 14-3. Listening to repeats of phase pattern.

```
cm> (events (piano2 pp-trope 1 1 .5 0) "reich.midi")
"reich-2.midi"
cm>
```

- reich-2.midi

## Opus 5, No. 1: The final version

All that remains is to define a wrapper function called **`piano-phase`** to create both processes with one call and to reimplement our **`piano1`** function to also accept the *stay* and *move* repetition factors for the pattern. The separate *stay* and *move* values will also allow us to experiment with different phasing proportions without having to redefine the functions each time.

Example 14-4. The final version of piano1 and the pphase function.

```
(define (piano1 trope stay move amp chan)
  (let* ((tlen (length trope))
         (cycl tlen)
         (reps (* tlen (+ stay move) cycl))
         (coda (* stay tlen))
         (rate (rhythm->seconds pp-pulse pp-tempo)))
    (process repeat (+ reps coda)
             for i from 0
             for x = (mod i tlen)
             for k = (list-ref trope x)
             output (new midi :time (now)
                          :keynum k
                          :duration (* rate 1.5)
                          :amplitude amp
                          :channel chan)
             wait rate)))

(define (pphase trope stay move amp)
  (list (piano1 trope stay move amp 0)
        (piano2 trope stay move amp 1)))
```

Before listening to the next phasing examples make sure that you have piano sounds selected on the first two channels of your synthesizer. The first interaction in Interaction 14-4 creates a score with *stay* and *move* both 1. This produces a very short version of the piece (about 40 seconds) in which the short phasing sections sound more like cross rhythms than they do actual phasing. The second score keeps *stay* at 1 repetition but increases *move* to 8, which allows the phasing sections to evolve at a more realistic rate of eight cycles of the trope. This second version of the piece lasts 3 minutes.

Interaction 14-4. Listening to pphase.

```
cm> (events (pphase pp-trope 1 1 .5) "reich.midi")
"reich-3.midi"
cm> (events (pphase pp-trope 1 8 .5) "reich.midi")
"reich-4.midi"
cm>
```

- reich-3.midi
- reich-4.midi

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file reich.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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H. Taube
06 May 2003

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# 15 Microtonality, Tunings and Modes

There is still a lot of good music waiting to be written in C major.

— Arnold Schönberg

One of the most liberating aspects about using a computer to compose music is that any frequency can be addressed without regards to its playability on a physical instrument or its inclusion in any particular scale. In this chapter we explore microtonality and "alternate" tuning systems, which we define very broadly to mean any tuning system other than the standard 12-tone equal tempered system.

## MIDI and Microtonal Tuning

Working with an expanded frequency space quickly exposes some weaknesses in the MIDI protocol. MIDI was developed by instrument manufacturers to serve the needs of the popular music industry. As a result, the protocol inherently reflects many of the characteristics of the Western tonal system. Nowhere is this bias more pronounced than in the implementation of frequency control. MIDI Note-On and Note-Off messages quantize an infinite number of frequencies into 128 *key numbers* that collectively represent a modern keyboard, and by extension, the 12-tone equal tempered tuning system to which the keyboards sold by music instrument manufacturers adhere to. This bias means that composers working outside the popular, mainstream tradition must overcome a built-in bias of MIDI synthesizers and MIDI messages toward the standard chromatic scale.

There are two basic methods to produce microtones on MIDI synthesizers. The first method is to retune the synthesizer by hand and then activate the retuned sounds using normal key number data. The advantage with this approach is that the stream of messages between the computer and synthesizer does not have to change, i.e. nothing extra needs to be done to produce microtones. A few disadvantages with this method are: (1) tuning methods vary with each synthesizer and so cannot be explained in a general way, (2) a particular manner of retuning may not be supported by the synthesizer, (3) retuning may globally effect all sound in the synthesizer, and (4) the retuning may be difficult and time consuming to do by hand.

An alternate approach to producing microtones on a synthesizer is to include specific tuning instructions in the stream of messages sent to the synthesizer. This kind of tuning is called *channel tuning* because it only affects sounds on specific channels of the synthesizer rather than globally in the synthesizer. The instructions that are used to make microtonal frequency adjustments are called *pitch bend messages*. The effect of these messages can heard by moving the pitch bend wheel on a MIDI keyboard to bend a sounding tone upward or downward by some amount. On many synthesizers the maximum pitch bend width is set to a major second on either side of a key number. This value is important to know because the tuning instructions sent to the synthesizer are defined as a "percentage" of this value. If the maximum pitch bend value is not a major second, consult the explanation of **`:pitch-bend-width`** in Chapter 11.

## Channel Tuning

Channel tuning works by determining the closest integer key number to a desired untempered frequency and then calculating a specific pitch bend adjustment to retune the integer key number by the appropriate microtonal amount. The pitch bend instruction is sent to the synthesizer immediately before the Note-On containing the integer key number is sent. The advantages that channel tuning has over retuning the synthesizer by hand are that any tuning adjustment can be made and these adjustments happen automatically in software rather than through the labor of the composer. The disadvantage with channel tuning is that MIDI pitch bends are *channel messages* that affect all sounds on a particular channel, not just specific key numbers. This means that if a sound is currently playing on the same channel to which a pitch bend is sent, the existing sound will be affected as well as the tone for which the pitch bend is intended. The basic strategy for channel tuning, then, is to send bend and key number pairs to channels that are currently silent. Since this routing will happen automatically, this generally requires that the composer allocate some range of channels for the software to maintain as a "pool" of available channels.

## Note by note tuning

There are several different strategies for managing this pool of tuning channels. In *note-by-note tuning* each pitch bend and key number pair is sent to the next available channel in the pool. The selection of this channel can be made either by maintaining a queue of available "open" channels (channels that are not currently sounding a note) or simply by cycling thorough a sufficiently large number of channels such that no overlapping notes in any channel will be produced. Both queuing and cycling methods send an exact microtonal pitch bend adjustment for each note. The disadvantage with both methods is that it is possible to run out of available channels if there are too many overlapping durations in the MIDI data.

## Equal division tuning

An alternate method for implementing channel tuning is to quantize microtones to a sufficiently small division of the semitone such that one gets close enough to the desired microtonal inflection. If equal division is used then a set of channels can be pretuned to a specific quantization size. This means that pitch bends do not need to be sent on each note and so a pitch bend cannot incorrectly affect overlapping sounds. The disadvantage, of course, is that the quantization itself may affect the perception of microtonal inflections. The minimum possible quantization step size in Common Music is 6.25 cents, very close to the smallest perceivable change in frequency by humans. Only experimentation can determine the best channel tuning strategy for a given situation. If quantization is not an issue — for example the microtones are equal tempered, or the texture is sufficiently dense so that it does not matter — then the quantization is inherently more efficient and less problematic than the note-by-note channel tuning method.

## Channel Tuning in Common Music

Common Music supports both note-by-note and equal division styles of channel tuning. Either method can be specified using the **`:channel-tuning`** keyword argument to the **`events`** function. The examples in this chapter tend to use the quantization method with semitones quantized to 9 equal microtonal divisions, approximately 11 cents per quantization step. Channel tuning will use the first nine channels in order to avoid the drum track channel that is hardwired in some synthesizers. You should experiment with different forms of channel tuning to compare the results. The documentation on MIDI files in Chapter 11 contains complete information on the **`:channel-tuning`** initialization to MIDI files. This documentation should be reviewed before beginning to work with the examples.

# The Harmonic Series

In Chapter 10 we learned that most musical instruments produce complex wave forms whose frequencies are related by integer ratios in the *harmonic series* (Figure 1). The harmonic series can be generated from any frequency *f* by simple multiplication: *f*1*, *f*2*, *f*3...f*n*. The first harmonic *f*1* is called the *fundamental*, the rest of the frequencies in the series are called *overtones*. A loop can be used to generate this series based on any fundamental *f*:

Interaction 15-1. Computing a harmonic series based on 100 Hz.

```
cm> (loop with f = 100 for h from 1 to 12
          collect (* f h))
(100 200 300 400 500 600 700 800 900 1000 1100 1200)
cm>
```

Since the series involves iteration we can also define a process to play any segment of the harmonic series between a lower and upper harmonic number:

Example 15-1. Playing the harmonic series.

```
(define (play-harms frq low high rate)
  (process for h from low to high
           for f = (* frq (/ h low))
           for k = (keynum f :hz)
           output (new midi :time (now)
                        :keynum k
                        :duration (* rate .9))
           wait rate))
```

The **play-harms** process produces tones in the harmonic series between a *low* and *high* harmonic number based on a low frequency *frq*. The frequency *f* of each harmonic *h* is determined by multiplying *frq* by the ratio *h/low* so that the series starts with *frq* as the *low* harmonic. The **keynum** function returns a floating point MIDI key number *k* for each Hertz value *f*. The **:hz** keyword after *h* in the argument list tell **keynum** that *f* is a Hertz number.

Interaction 15-2. Playing the harmonic series.

```
cm> (events (play-harms 440 8 16 .4) "micro.midi"
            :channel-tuning true)
"micro-1.midi"
cm> (events (play-harms 440 8 16 .4) "micro.midi"
            :channel-tuning 9)
"micro-2.midi"
cm> (events (play-harms 440 8 16 .4) "micro.midi"
            :channel-tuning false)
"micro-3.midi"
cm>
```

- micro-1.midi
- micro-2.midi
- micro-3.midi

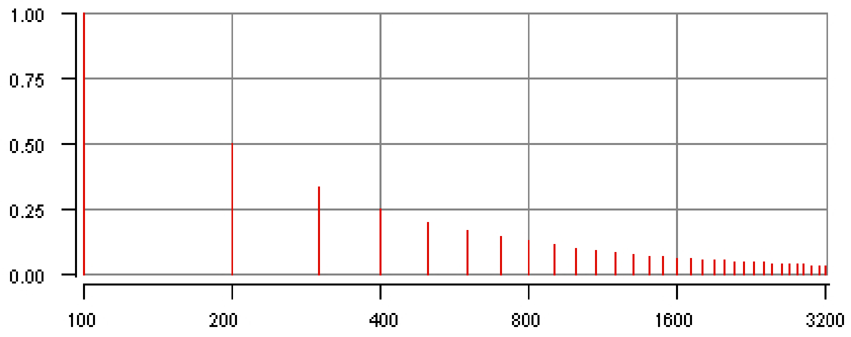
The last events call in Interaction 15-2 "turns off" channel tuning. As an alternate method for canceling pitch bend effect one could define a process that sends a series of level 0 pitch bends to every channel to reset the synthesizer to its "normal" state:

Example 15-2. Clearing pitchbends.

```
(define (clearbends )
  (process for c below 16
           output (new midimsg :time (now)
                        :msg (make-pitch-bend c 0))))
```

One of the most interesting properties of the harmonic series is its fractal nature: each prime number in the series introduces a new (non-doubled) tone into the series that then becomes a “new fundamental” for generating another harmonic series. For example, the third harmonic in Example 2 at 300 Hz is the first frequency in the series that is not a doubling (octave) of the fundamental and generates its own series: 300, 600, 900, 1200 and so on. Similarly, the 5th harmonic generates the series: 500, 1000, 1500... Since the harmonic series is linear but octaves are powers of two there are more and more harmonics per octave as the series rises: the first octave in the series contains 1 frequency, the second 2, the third 4, the fourth 8 and so on. We do not hear the increasing harmonic density as harshness in the timbre of musical instruments because the amplitudes of the higher harmonics tend to fall off in a linear way with the harmonic number so the higher, quieter harmonics quickly become *masked* by the louder, more widely spaced frequencies lower in the spectrum (Figure 1).

Figure 15-1. The first 32 harmonics in the harmonic series based on a fundamental of 100 Hz. The amplitude of each harmonic has been calculated as the reciprocal of the harmonic number, or *1/h*.



We can hear the relationship between higher harmonics by defining a process to play the series wrapped to fit within a single octave:

Example 15-3. Playing the harmonic series.

```
(define (play-harms2 octs)
  (process with hz = (hertz 'c3)
           for h from 1 to (* 12 octs)
           for k = (keynum (* h hz) :hz)
           output (new midi time (now)
                        duration 1.0
                        amplitude (between 0.4 0.6)
                        keynum (+ 48 (mod k 12)))
           wait .15))
```

The **`play-harms2`** process plays the harmonic series based on the hertz value C3 in the standard chromatic scale. The series is produced modulo the octave (12 semitones) so that each consecutive octave in the series starts on the same bottom tone. Notice that each time the process starts over on the low note the number of divisions per octave doubles.

Interaction 15-3. Playing four octaves of a normalized harmonic series.

```
cm> (events (play-harms2 4) "harms.midi" :channel-tuning 9)
"harms-1.midi"
cm> (events (clearbends ) "clearbends.midi")
"clearbends-1.midi"
cm>
```

- harms-1.midi
- clearbends-1.midi

## Intervals in the Harmonic Series

Since each ascending octave in the series contains more frequencies, it stands to reason that the intervals between consecutive harmonics in the series decrease as the harmonic number increases. Musically speaking, these intervals are a difference — the space between two tones. But physically speaking, an interval is a proportion, or ratio, between two frequencies. Ascending or descending intervals between any two tones in the harmonic series can be determined by the ratio of their harmonics numbers. The order in which the ratio is expressed does not matter — if the ratio is larger than 1 then an ascending interval is described, if the ratio is less than 1 then the interval is downward. For example, a 2/1 ratio produces an octave up and its reciprocal 1/2 is an octave down. Similarly, 3/2 is a perfect fifth up and 2/3 is a perfect fifth down. The 3/2 perfect fifth is called *just interval* because it is found in the harmonic series, i.e. it is a ratio of two whole numbers. In contrast, the ratio for an equal-tempered perfect 5th is $2^{7/12}$, or 1.4983070768766815, which is very close to, but not equal with 1.5 or 3/2. Ratios for some common just musical intervals are shown in Table 1.

Table 15-1. Ratio and cent values of common just intervals.

| interval | ratio | cents | interval | ratio | cents |
|---|---|---|---|---|---|
| unison | 1/1 | 0 | perfect 5th | 3/2 | 702 |
| minor 2nd | 16/15 | 112 | minor 6th | 8/5 | 814 |
| major 2nd | 9/8 | 204 | major 6th | 5/3 | 884 |
| minor 3rd | 6/5 | 316 | minor 7th | 9/5 | 1018 |
| major 3rd | 5/4 | 386 | major 7th | 15/8 | 1088 |
| perfect 4th | 4/3 | 498 | octave | 2/1 | 1200 |
| tritone | 45/32 | 590 | | | |

## Interval Math

Since intervals are ratios they must be combined using multiplication and division rather than addition and subtraction. For example, to *add* the intervals of a just fifth and fourth, multiply their ratios together:

(3/2 * 4/3) = 12/6 = 2/1

Similarly, the “subtraction” of minor third from an octave requires division:

(2/1 / 6/5) = (2/1 * 5/6) = 10/6 = 5/3

To invert a perfect fifth means to take its reciprocal by dividing into 1.

(1/1 / 3/2) = (1/1 * 2/3) = 2/3

To find the complement of a ratio, multiply the smaller of the numerator or denominator by 2. If the numerator is smaller then multiply by 2/1, if the denominator is smaller multiply by 1/2. For example, to find the complement of 3/2 multiply by 1/2 to produce the perfect fourth:

(3/2 * 1/2) = 3/4 = 4/3

and a just major sixth is the complement of a minor third:

(5/6 * 2/1) = 10/6 = 5/3 = 3/5

## Cents

Sometimes the proportion that a ratio describes may be non-intuitive. For example, which is larger: 256/243 or 9/8? The *cent scale* was devised as an alternative method for describing the distance between two frequencies. There are 100 cents in a tempered semitone and 1200 cents per octave. One cent is 1/1200 of an octave — an imperceptibly small change in frequency. The smallest frequency difference a human can perceive is approximately 5 cents so the cent scale is more than adequate for describing minute changes in frequency. Unlike ratios, cent measurements can be combined using addition and subtraction. For example the size of just perfect fifth can be determined by adding the cent values of a just major third and a just minor third: 316+386=702 cents. Common Music provides two functions that convert between cents and ratio scalers:

[Function]
(**cents->scaler** *cents*)
[Function]
(**scaler->cents** *ratio*)

The **cents->scaler** function converts *cents* into the scaler value $2^{cents/1200}$ while the **scaler->cents** function performs the reverse calculation of $1200*\log_2 ratio$.

Interaction 15-4. Converting cents to scaler values.

```
cm> (cents->scaler 1200)
2.0
cm> (scaler->cents 3/2)
702
cm> (scaler->cents 5/4)
386
cm> (scaler->cents 6/5)
316
cm> (cents->scaler 700)
1.4983070768766815
cm>
```

## Listening to Microtonal Intervals

We can define functions that create MIDI events to play and compare microtonal intervals:

Example 15-4. Playing microtonal intervals.

```
(define (midi-ratio at note ratio dur)
  (let ((fkey (keynum (* (hertz note) ratio) :hz)))
    (list (new midi :time at :keynum note :duration dur)
          (new midi :time at :keynum fkey :duration dur))))

(define (compare-ratios note rat1 rat2 dur)
  (append (midi-ratio 0 note rat1 dur)
          (midi-ratio (+ dur .5) note rat2 dur)))
```

```
(define ct8 (list true 1 9))
```

The **`midi-ratio`** function returns two midi events a *ratio* interval apart. The first event in the list plays the specified *note* in the standard chromatic scale. The second event plays a tone *fkey* calculated to be *ratio* distance above or below *note*. This value is determined by multiplying the hertz value of *note* by *ratio* and then converting the value to a floating point key number in the chromatic scale. When channel-tuning is enabled the second tone will be heard with its microtonal inflection.

The second function **`compare-ratios`** allows us to compare two different intervals. This function simply appends the results of two calls to **`midi-ratio`**. The second interval will be played .5 seconds after the first interval ends. The **`ct8`** variable defines a **`:channel-tuning`** value for note-by-note tuning using channels 1-8. This keeps channel 0 free of pitch bends for the first tone of each interval, which does not have a floating point key number and should not be retuned.

Interaction 15-5. Playing and comparing ratios.

```
cm> (events (midi-ratio 0 'c4 6/5 3) "rat.midi"
            :channel-tuning true)
"rat-1.midi"
cm> (events (compare-ratios 'c4 6/5 (expt 2 4/12) 3)
            "rat.midi" :channel-tuning ct8)
"rat-2.midi"
cm>
```

- rat-1.midi
- rat-2.midi

In the first input we listen to a just major third. The second example compares the just major third with the equal tempered major third of $2^{4/12}$. The just major third is 386 cents, 86 cents wider than the equal tempered major third.

## Scales and Temperament

A *scale* quantizes infinite frequency space into a finite number of steps, called *scale degrees*. The scale degrees are generally ordered low to high and serve as the raw material for a composition. The term *temperament* refers to the precise relationships between the degrees in a scale, and a *tuning system*, describes a formal method to temper each degree.

### Scales and Tunings In Common Music

A scale in Common Music is an object that defines a series of steps called scale degrees. Each scale degree is represented by an integer position in the scale called a *keynum*, a hertz value, and possibly by one or more *note names* associated with each degree. The term *scale* in Common Music applies to two different types of objects. A **`tuning`** is a scale that defines the precise frequencies for between scale degrees. A **`mode`** defines a transposable subset of a **`tuning`**. Modes are discussed in a later section of this chapter.

[Class]
  **`tuning`**

A **`tuning`** is a scale with Hertz frequencies defined for each scale degree. Scale degrees may also have note names associated with them.

**`tuning`** supports the following slot initializations:

:name {*string* | *symbol*}
: An optional name for the tuning.

:lowest *number*
: The lowest Hertz value defined in the scale. Defaults to 8.175, which is C-1 in the standard chromatic scale.

:cents *list*
: One octave of the tuning specified as a list of cent values. Fewer or more than 12 degrees may be specified and an "octave" does not have to equal 1200 cents. If the cent values in *list* start with 0 and are in increasing order they specify the cent difference from the lowest degree to each degree above it. In this case the last value in *list* determines the octave width of the scale. If *list* does not begin with 0 then the cents values describe the distances between adjacent steps. In this case the octave width is set to the sum of the cent steps. To define a tuning with note names specify each step as a list: `(`*cent* `{`*note*`}*)` where *cent* is the cent value followed by one or more *note* declarations. Each *note* declaration may consist of symbol *note* or a list `(`*note* `&key :accidental)` where the value of **`:accidental`** is the portion of *note* that represents the accidental. For example, the 8th step in the definition of the standard chromatic scale is:

```
        (100 (af :accidental f) (gs :accidental s))
```

:ratios *list*
: Exactly like **`cents`** except that the tuning values are specified as ratios (direct frequency scalers) rather than as cent values.

:octaves {*number* | *list*}
: The number of octaves defined in the scale. Defaults to 10. The value can be a number or a list of two values **(***start end***)** where *start* is the first octave number and *end* is the last octave number. Specify false if the tuning has no octaves.

:keynum-offset *integer*
: A integer offset to add to all keynums values returned by **`keynum`**. Defaults to 0.

:default-octave *integer*
: The default octave for notes without octave numbers. Defaults to the 4th octave.

## Tuning Systems

In the remaining sections of this chapter we present some examples of tuning systems that demonstrate different approaches to temperament. There is really no way to adequately cover the full breadth and variety of different tuning systems in a single chapter. The Web is an excellent resource for gathering descriptions of tuning systems; many of these descriptions provide ratios or cent specifications and can be easily implemented by tuning objects in Common Music. Searching for Alternate Tunings and Tuning Systems in Google will give you a good candidate list to start exploring!

Before presenting the example tunings we first define a process to play the examples. The process will preview scale degrees of a specified tuning in ascending, descending, and random order. This allows the tonal material of the scale to be heard in several different relationships.

Example 15-5. A process to play tuning examples.

```
(define (play-tuning tuning low high rate amp)
  (let* ((rise (loop for i from low below high collect i))
         (fall (loop for i downfrom high downto low collect i))
         (ran1 (shuffle (rest rise)))
         (ran2 (shuffle (rest rise)))
         (coda (loop for i from low to high collect i))
         (all (append rise fall ran1 ran2 coda)))
    (process for k in all
             for x = (keynum k :from tuning)
             output (new midi :time (now)
                          :keynum x
                          :amplitude amp
                          :duration (* rate .95))
             wait rate)))
```

The **play-tuning** process previews the scale degrees of *tuning* between a *low* and *high* key number. The variables *rise* and *fall* contain the ascending and descending patterns, while *ran1* and *ran2* hold randomized versions of this material. (Common Music's **shuffle** function randomly reorders lists of data, and is discussed in Chapter 18) The *coda* segment of the melodic material is again a step wise ascent from *low* and *all*. *All* holds the entire list of key numbers to play created by appending all of the sublists together into a single long list of key numbers. The variable *x* holds a key number that has been converted from the tuning to an equivalent value in the standard chromatic scale. The **:from** argument to **keynum** converts an integer key number from the specified tuning object into an equivalent floating point key number in the standard chromatic scale.

## The Harmonic Scale

We can treat a segment of the harmonic series as a scale by defining a function that converts harmonic numbers into ratios. These ratios can then be used **:ratios** values for a tuning.

Example 15-6. Converting harmonic numbers to ratios.

```
(define (harms->ratios low high)
  (loop for h from low to high collect (/ h low)))

(new tuning :name 'jmaj
     :ratios (harms->ratios 8 16))

(new tuning :name 'jmin
     :ratios (harms->ratios 9 18))
```

In the first scale we use an 8 note segment of the harmonic series from the 8th to the 16th partial to give us a just major scale, the second scale defines a nine tone octave based on the 9th partial with a just minor third between the 1st and 3rd scale degrees. Both tunings are based on a low C (8.125 Hz). This means that Middle C is key number 40 in the Jmaj scale and 45 in the jMin tuning.

Interaction 15-6. A process to play microtonal examples.

```
cm> (hertz 40 :in #&jmaj)
261.6255653005986
cm> (hertz 45 :in #&jmin)
261.6255653005986
cm> (keynum 40 :from #&jmaj)
60.0
cm> (keynum 41 :from #&jmaj)
62.039100017307746
cm> (events (play-tuning #&jmaj 40 56 .25 .6)
            "hscale.midi" :channel-tuning 9)
"hscale-1.midi"
cm> (events (play-tuning #&jmin 45 63 .25 .6)
            "hscale.midi" :channel-tuning 9)
```

```
"hscale-2.midi"
cm>
```

- hscale-1.midi
- hscale-2.midi

## Pythagorean Tuning

The Pythagorean Scale, named after the Greek mathematician and philosopher Pythagoras of Samos who discovered it, was the primary tuning system in Western music for about 2000 years. Pythagoras noticed a pleasing scale could be built using a 3/2 perfect fifth between each tone. Rather than work with steps that are all 3/2 apart, the ratios in the Pythagorean scale are normalized to a single octave by reducing any ratio greater than 2 (the octave) by 1/2 so that the value fits within a single octave. For example, the first tone in the scale (call it C) has a ratio of 1/1 = 1. The second tone G is (1 * 3/2) = 3/2; the third tone D is (3/2 * 3/2 * 1/2) = 9/8; A is 9/8 * 3/2 = 27/16 and so on. The tones are then ordered to form an ascending step wise pattern from C and the missing F is calculated downward from C using 4/3, the interval complement of 3/2. Table 2 shows the ration of the seven note Pythagorean Diatonic Scale:

Table 15-2. Step ratios in the diatonic Pythagorean scale.

| C | D | E | F | G | A | B | C |
|---|---|---|---|---|---|---|---|
| 1 | 9/8 | 81/64 | 4/3 | 3/2 | 27/16 | 243/128 | 2 |

Given our set of ratios we can create a tuning to implement the Pythagorean diatonic scale:

Example 15-7. Tuning definition for the Pythagorean Diatonic Scale

```
(new tuning :name 'pyth-d
     :ratios '(1 9/8 81/64  4/3  3/2  27/16 243/128 2))
```

The tuning named pith-d in Example 15-7 is a seven tone per octave tuning based on a low C at 8.175 Hz. Since there are 7 notes per octave, Middle C is key number 35 in pyth-d, and D4, which is key number 62 in the standard chromatic scale, is key number 36 in the pyth-d scale.

Interaction 15-7. Converting key numbers to and from a tuning.

```
cm> (keynum 60 :to #&pyth-d)
35
cm> (keynum 70 :to #&pyth-d)
35BUG!!!!!!!
cm> (hertz 35 :in #&pyth-d)
261.6255653005986
cm> (keynum 35 :from #&pyth-d)
60.0
cm> (keynum 36 :from #&pyth-d)
62.039100017307746
cm>
```

We can now listen to the sound of the scale. The key numbers specified to **`play-tuning`** are in the coordinates of the pyth-d tuning system, the process converts these to floating point equivalents in the standard scale.

Interaction 15-8. Playing the Pythagorean diatonic scale.

```
cm> (events (play-tuning #&pyth-d 35 49 .3 .7) "pyth.midi"
            :channel-tuning 9)
"pyth-1.midi"
cm>
```

- pyth-1.midi

## The Pythagorean Chromatic Scale

If we continue the 3/2 ratio series from Table 2 we can eventually generate all 12 pitches in the cycle of 5ths (Table 3):

Table 15-3. Fifth ratios in the chromatic Pythagorean scale.

| | | | | | |
|---|---|---|---|---|---|
| F | $3/2^{-1}$ | 2/3 | F# | $3/2^{6}$ | 729/64 |
| C | $3/2^{0}$ | 1 | C# | $3/2^{7}$ | 2187/128 |
| G | $3/2^{1}$ | 3/2 | G# | $3/2^{8}$ | 6561/256 |
| D | $3/2^{2}$ | 9/4 | D# | $3/2^{9}$ | 19683/512 |
| A | $3/2^{3}$ | 27/8 | A# | $3/2^{10}$ | 59049/1024 |
| E | $3/2^{4}$ | 81/16 | E#/F | $3/2^{11}$ | 177147/2048 |
| B | $3/2^{5}$ | 243/32 | B#/C | $3/2^{12}$ | 531441/4096 |

The *circle of 5ths* tells us that after a series of 12 5ths we should end up on a power of two, some octave doubling of our original frequency. In other words, for the circle of 5ths to be a true circle, $3/2^{12}$ must equal to $2^{7}$. As we can plainly see in Table 15-3 this is not true!

Interaction 15-9. The Big Lie.

```
cm> (expt 3/2 12)
531441/4096
cm> (expt 2 7)
128
cm> (= 531441/4096 128)
#f
cm> (/ 531441/4096 128.0)
1.0136432647705078
cm> (scaler->cents 1.0136432647705078)
23
cm>
```

The difference between $3/2^{12}$ and $2^{7}$ is 1.746337890625 or 23 cents. So the “circle of fifths” is fiction, at least if one uses the 3/2 fifths that nature gives us. The 23 cent difference between the octaves and fifths is called the *Pythagorean comma* and it is one of the central issues in the design of tuning systems that followed the Pythagorean Scale. The discrepancy between the cycle of fifths and octaves means that it is mathematically impossible to design a tuning system in which all intervals are just while still remaining true to the octave.

There are other problems lurking in the Pythagorean tuning as well. While fifths and whole steps are just, both the major and minor thirds are out of tune. The Pythagorean major third is 81/64 (408 cents), 22 cents wider then the just major third, and the Pythagorean minor third 32/27 (294 cents) is 22 cents smaller than the just minor third. This 22 cent difference between Pythagorean and just thirds is called the *syntonic comma*. Half steps in the Pythagorean scale have a different problem. Notice that the half step ratio between E (81/64) and F (4/3) is (4/5*64/81=256/243, or 90 cents. But the ratio between F and F-sharp is 2187/2048, or 114 cents! This means that the chromatic Pythagorean scale has two versions of the half step: 90 cents and 114 cents.

Interaction 15-10. Comparing major thirds, minor thirds and minor seconds.

```
cm> (events (compare-ratios 'c4 5/4 81/64 3)
            "maj3.midi" :channel-tuning '(#t 1 3))
"maj3-1.midi"
cm> (events (compare-ratios 'c4 6/5 81/64 3)
            "min3.midi"
            :channel-tuning '(#t 1 3))
"min3-1.midi"
cm> (events (compare-ratios 'c4 256/243 2187/2048 3)
            "min2.midi"
            :channel-tuning '(#t 1 3))
"min2-1.midi"
cm>
```

- maj3-1.midi
- min3-1.midi
- min2-1.midi

We can now define a version of the Pythagorean chromatic scale whose degrees are expressed by the cent measurements between its minor seconds:

Example 15-8. Pythagorean chromatic scale.

```
(new tuning :name 'pyth-c
     :cents '(114 90 90 114 90 114 90 114 90 90 114 90))
```

Interaction 15-11. Listening to the Pythagorean chromatic scale.

```
cm> (events (play-tuning #&pyth-c 60 84 .3 .7)
            "pyth.midi" :channel-tuning 9)
"pyth-2.midi"
cm>
```

- pyth-2.midi

## Equal Temperament

In the Pythagorean scale fifths and whole steps are just but thirds are badly out of tune. As the role of the third became more important in Western music, composers searched for alternate tuning methods to replace the Pythagorean system. Different tunings arose, in part, because there are different ways to deal with the comma differences exhibited in the Pythagorean scale. *Quarter Comma Meantone Tuning* distributes the syntonic comma equally over four 5ths: C G D A, which allows the thirds within the fifths to remain just. But of course, each fifth is then out of tune at 696.5 cents, or 1/4 syntonic comma (22/4 cents) flatter than the just fifth. Quarter Comma Meantone Tuning allows closely related keys to sound in tune while distant keys are badly out of tune. *Just Tuning* tunes triads — a series of stacked major and minor triads are justly tuned but more distantly related triads are out of tune. *12-tone equal temperament* takes a different approach altogether. Rather than keeping some ratios just at the expense of others, equal temperament distributes the syntonic comma equally over all 12 fifths, thereby reducing each just fifth by 2 cents to 700 cents. This means that in equal temperament there are no just intervals at all, but all intervals are affected equally by only a slight amount. In Chapter 7 we learned that another way to think about equal temperament is as a system that divides the octave into $n$ equal half steps with a ratio size of $2^{1/n}$ between each step. In the case of the standard chromatic scale $n$ is 12 and the step ratio is $2^{1/12}$, or 1.0594630943592953. But of course there is nothing magical about 12 steps, we can see in Example 15-9 that it is just as easy to design tuning systems with more or fewer steps, or to base the equal temperament on an octave size that is not a power of two.

Example 15-9. Alternate equal tempered scales.

```
(new tuning :name '5-tone
     :cents (loop repeat 5 collect (/  1200 5)))

(new tuning :name '24-tone
     :cents (loop repeat 24 collect 50))

(new tuning :name 'bohlen-pierce
     :ratios (loop for i to 13 collect (expt 3.0 (/ i 13))))
```

The 5-tone equal tempered scale is a very rough approximation of the Javanese Slendro scale (which has no single exact tuning). 24-tone implements quarter tone tuning. The Bohlen-Pierce scale is a 13-step scale with an octave ratio of 3/1 rather than 2/1. In other words, the “octave” for the Bohlen-Pierce scale is the octave-plus-a-fifth interval between the first and third harmonics in the harmonic series. There are also justly tuned chromatic and diatonic versions of the Bohlen-Pierce scale. For more information on the Bohlen-Pierce scale, consult Current Directions in Computer Music Research, Mathews and Pierce editors.

## Pelog

Pelog is a seven-tone, non-octave scale characteristic of Javanese gamelan music. The degrees in the Pelog scale are named but there is no single relationship defined between them. Instead, each gamelan is tuned to its own version of Pelog. The two “octave” version shown in Example 15-10 was taken from [Harvard Dictionary of Music].

Example 15-10. A two octave Pelog scale.

```
(new tuning :name 'pelog
     :lowest 220.0
     :octaves #f
     :cents '((0    nem0)
              (125  barang0)
              (266  bem0)
              (563  gulu0)
              (676  dada0)
              (800  pelog0)
              (965  lima0)

              (1220 nem1)
              (1360 barang1)
              (1503 bem1)
              (1778 gulu1)
              (1905 dada1)
              (2021 pelog1)
              (2225 lima1)

              (2447 nem2)))
```

## La Monte Young's Well Tuned Piano

La Mont Young's piano composition *The Well Tuned Piano* is based on a 12-tone, justly tuned scale that features the “flat” 7th partial in its design. A complete description of the scale can be found in La Monte Young's The Well-Tuned Piano by Kyle Gann. Young's scale (Table 15-4) is a *seven limit* tuning, which means that the ratios defining the scale degrees contain no prime numbers greater than seven.

Table 15-4. Step ratios in Young's scale.

| Eb | E | F | F# | G# | G | A |
|---|---|---|---|---|---|---|
| 1/1 | 567/512 | 9/8 | 147/128 | 1323/1024 | 21/16 | 189/128 |

| Bb | B | C# | C | D | Eb |
|---|---|---|---|---|---|
| 3/2 | 49/32 | 441/256 | 7/4 | 63/32 | 2 |

This tuning has a number of interesting features. First, it contains a wide variety of interval sizes, ranging from the smallest, a 27 cents half-step between E and F, up to 204 cents, the just major second between E-flat and F. The scale's steps are also grouped into a pentatonic clustering in which the groups are successively centered around E-flat, F, G, B-flat and C in the standard chromatic scale. Another interesting feature of the scale is its use of ratios involving the “flat” seventh harmonic to establish consonant thirds relations. Young's tuning uses a 9/7 major third between E and G-sharp, G and B, A and C-sharp, D and F-sharp, and a 7/6 minor third between F-sharp and A, B and D, and C-sharp and E. Another unusual feature of the scale is that its G-sharp is lower than its G, and C-sharp is is lower than C. This reordering was done to create 3/2 perfect fifths between the F-sharp, C-sharp and G-sharp spellings.

Example 15-11. Young's scale.

```
(new tuning :name 'young
     :ratios '((1 ef)
               (567/512 e)
               (9/8 f)
               (147/128 fs)
               (1323/1024 gs)
               (21/16 g)
               (189/128 a)
               (3/2 bf)
               (49/32 b)
               (441/256 cs)
               (7/4 c)
               (63/32 d)
              2)
     :octaves '(-1 10)
     :lowest (* (hertz 'a-1) 128/189))
```

Young's scale is based on an E-flat that is actually closer to D than it is to E-flat in the standard chromatic scale. But the sixth scale degree, A, is tuned to 440.0 Hz, which means that the scale's lowest E-flat can be calculated by multiplying the lowest A in the standard chromatic scale by the reciprocal of the sixth degree's ratio of 128/189. The scale degree ratios in Interaction 15-12 are listed in monotonically increasing order, which means that the symbols that define the G-sharp and C-sharp note names appear before their natural versions. The last ratio in the list is 2, which establishes the octave ratio without defining a redundant note name.

Interaction 15-12. Values in Young's scale.

```
cm> (note 0 :in #&young)
ef-1
cm> (hertz 'a4 :in #&young)
440.0
cm> (keynum 'c4 :in #&young)
70
cm> (hertz 'c4 :in #&young)
```

```
521.4814814814814
cm>
```

The following musical examples were implemented by Michael Klingbeil to demonstrate Young's tuning.

Example 15-12. A process to preview Young's scale.

```
(define (arp-chord notes rate dur amp)
  ;; arpeggiate a list of notes.
  (process with transp = 0
           for n in notes
           for k = (keynum n :from #&young)
           output (new midi :time (now)
                        :keynum (+ transp k)
                        :duration dur
                        :amplitude amp)
           wait rate))
```

The following two examples demonstrate the basic scaler and intervalic relations in Young's scale.

Example 15-13. Scale and septimal interval relations.

```
(define youngs-scale
   '(ef4 e4 f4 fs4 gs4 g4 a4 bf4 b4 cs5 c5 d5 ef5))
(define sept-7th         ; 7/4 seventh
   '(ef4 c5))
(define sept-min-3rds    ; 7/6 thirds
   '(e4 gs4 f4 g4 a4 cs5))
(define sept-maj-3rds    ; 9/7 thirds
   '(fs4 a4 b4 d5 cs5 e5 c5 f5))
```

Interaction 15-13. Listening to the scale and intervals.

```
cm> (events (arp-chord youngs-scale 0.4 2 .6)
            "young.midi" :channel-tuning true)
"young-1.midi"
cm> (events (play-tuning #&young 48 96 .15 .5)
            "young.midi")
"young-2.midi"
cm> (events (arp-chord sept-7th 0 3 .6)
            "young.midi")
"young-3.midi"
cm> (events (arp-chord sept-min-3rds 0 2 .6)
            "young.midi")
"young-4.midi"
cm> (events (arp-chord sept-maj-3rds 0 2 .6)
            "young.midi")
"young-5.midi"
cm>
```

- young-1.midi
- young-2.midi
- young-3.midi
- young-4.midi
- young-5.midi

Example 15-14 defines some representative sonorities that Young uses in The Well Tuned Piano.

Example 15-14. Chords from Well Tuned Piano.

```
(define opening-chord
   '(ef3 bf3 c4 ef4 f4 bf4))
(define magic-chord
   '(e3 fs3 a3 b3 d4 e4 g4 a4))
(define gamelan-chord
   '(fs3 a3 c4 e4))
(define tamiar-dream-chord
   '(b2 d3 g3 a3 b3 d4 g4 a4))
(define lost-ancestral-lake-region-chord
  '(g2 b2 d3 fs3 g3 a3))
(define brook-chord
   '(bf3 c4 ef4 f4 g4 bf4 c5 ef5 f5 bf5))
(define pool-chord
   '(ef3 f3 fs3 bf3 c4 ef4 f4 bf4))
```

Interaction 15-14 demonstrates the Opening Chord, Magic Chord, Gamelan Chord, Tamiar Dream Chord, 89/89 Lost Ancestral Lake Region, The Brook, and The Pool chords:

Interaction 15-14. Sonorities from The Well Tempered Piano.

```
cm> (events (arp-chord opening-chord 0.25 3 .6)
            "young.midi" :channel-tuning true)
"young-6.midi"
cm> (events (arp-chord magic-chord 0.25 3 .6)
            "young.midi")
"young-7.midi"
cm> (events (arp-chord gamelan-chord 0.25 3 .6)
            "young.midi")
"young-8.midi"
cm> (events (arp-chord tamiar-dream-chord 0.25 3 .6)
            "young.midi")
"young-9.midi"
cm> (events (arp-chord lost-ancestral-lake-region-chord
            0.25 3 .6)
            "young.midi")
"young-10.midi"
cm> (events (arp-chord brook-chord .15 5 .6)
            "young.midi")
"young-11.midi"
cm> (events (arp-chord pool-chord .15 5 .6)
            "young.midi")
"young-12.midi"
cm>
```

- young-6.midi
- young-7.midi
- young-8.midi
- young-9.midi
- young-10.midi
- young-11.midi
- young-12.midi

## Modes

A mode in Common Music is a scale that defines a transposable subset of a tuning. Adjacent degrees in the mode may be non-contiguous in the tuning.

[Class]
    **`mode`**

**`mode`** supports the following slot initializations:

:name {*string* | *symbol*}
    An optional name for the mode.
:tonic *note*
    Sets the tonic note name of the mode. Defaults to the first note in the steps specification, or C if the steps are described as intervals.
:scale *tuning*
    Sets the tuning system for the scale. Notes in the mode must also be notes in the tuning. Defaults to the standard chromatic scale.
:steps {*notes* | *intervals*}
    One octave of the mode specified as a list of notes or list of intervals between notes. If **`steps`** is a list of notes the tonic (transposition offset) of the mode is automatically set to the note class of the first note in the list. If the steps are specified as a list of intervals the transposition is set to C. A mode can be transposed to a new tonic using the **`transpose`** function.

Functions like **`hertz`**, **`keynum`**, **`note`** and **`transpose`** accept modes as well as tunings as scale inputs:

Interaction 15-15. Transposing modes.

```
cm> (new mode :name 'dorian :steps '(d e f g a b c d))
1
cm> (keynum 'd4 :in #&dorian)
35
cm> (loop for i from 35 to 42
          collect (note i :in #&dorian))
(d4 e4 f4 g4 a4 b4 c5 d5)
cm> (transpose #&dorian 'cs)
2
cm> (loop for i from 35 repeat 45
          collect (note i :in #&dorian))
(cs4 d4 e4 es4 fss4 gs4 as4 b4 cs5)
cm>
```

To demonstrate the use of modes we define a process that harmonizes random chords on Messiaen's Modes of Limited Transposition.

Example 15-15. Messiaen's Modes of Limited Transposition.

```
(new mode :name 'mode1 :steps '(c d e fs gs bf c))
(new mode :name 'mode2 :steps '(c df ef e fs g a bf c))
(new mode :name 'mode3 :steps '(c d ef e fs g af bf b c))
(new mode :name 'mode4 :steps '(c df d f fs g af b c))
(new mode :name 'mode5 :steps '(c df f fs g b c))
(new mode :name 'mode6 :steps '(c d ef fs gs as b c))
(new mode :name 'mode7 :steps '(c df d ef f fs g af a b c))
```

```
(define chords '((0 2 4 5) (0 1 3 5) (0 2 3 4)))

(define (messiaen mode start end rate)
  (let ((num (length chords)))
    (process for m from start to end
             for c = (list-ref chords (random num))
             for l = (note (transpose c m) :in mode)
             each n in l
             output (new midi :time (now)
                         :keynum n
                         :duration (* rate 2))
             wait (odds .15 (* rate .5) rate))))
```

The **`messiaen`** process selects chords at random from a list of possible chord templates defined in the global variable *chords*. Each chord template is a list of intervals that will be transposed to specific key numbers in the *mode* specified to the process. The **`wait`** expression uses Common Music's **`odds`** function to randomly produce “added value” rhythms that are characteristic of much of Messiaen's music.

Interaction 15-16. Listening to Messiaen's Modes

```
cm> (events (messiaen #&mode2 35 50 .4)
            "messiaen.midi")
"messiaen-1.midi"
cm> (events (messiaen #&mode6 35 50 .4)
             "messiaen.midi")
"messiaen-2.midi"
cm>
```

- messiaen-1.midi
- messiaen-2.midi

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file scales.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 16 Mapping

There is geometry in the humming of the strings.

— Pythagoras of Samos

Proportion, juxtaposition and balance between musical ideas are fundamental relations in music, and transformations that affect these properties permeate the craft of composition. Scaling and offsetting are two of the simplest yet compositionally most powerful procedures composers use to transform musical ideas. These operations are also very easy to implement in software. In this chapter we look at some computer techniques that are related to the operations of scaling and offsetting and demonstrate their control over the evolution of parameterized sound data.

## Scaling and Offsetting

As its name suggests, to *scale* a quantity means to change its size, or magnitude, through multiplication. In the visual domain, scaling causes an object to appear bigger or smaller without altering its origin with respect to the visual field. In music composition, the idea of scaling can be clearly observed operating in our system of mensural rhythmic proportions and in melodic inversion. Scaling is also evident in some types of formal augmentation and diminution, in the harmonic series and tuning systems, and in amplitude shapes such as the crescendo and decrescendo. To *offset* a quantity means to add a value to it. Offsetting affects the origin of something along an axis of interest. Offsetting can be observed in music as semitonal transpositions, used extensively in the 12-tone system. Rhythmic effects such as syncopation and the added values of Messiaen are also offset effects, as are repetition and sequencing of motives, patterns and other types of formal units.

### Mapping

Throughout history composers have shown a fascination for projecting extramusical ideas into sound. Dufay's famous isorhythmic motet *Nuper Rosarum Flores* was composed by incorporating architectural ratios of the dome of the cathedral of Santa Maria del Fiore in Florence onto musical proportions for his motet that was commissioned for the cathedral's opening in 1436. (see Charles Warren's Brunelleschi's Dome and Dufay's motet, Music Quarterly). Charles Dodge's composition *The Earth's Magnetic Field* is based on measurements of the Earth's yearly average magnetic activity incorporated onto a set of pitches for an eight minute piece. The general term for this type of transformation is called *mapping*. Mapping means to relate, or project, values from one domain of interest into another. We will implement a simple mapping function called **`rescale`** that uses only scaling and offsetting to transform values from one coordinate system into another.

## The Rescale Function

The **`rescale`** function implements a simple, general tool for effecting transformations by scaling and offsetting. The simplicity of the function belies its expressive power — we will have opportunities to use this function many times throughout the remained of this book to reformulate values from one range into another, or to translate a value in one parameter into a value for another parameter, or to project non-musical data into musical parameter space. The **`rescale`** function is defined by Common Music but is presented here in Example 16-1 to demonstrate how this type of scaling and offsetting can be implemented.

Example 16-1. Common Music's rescale function.

```
(define (rescale x old-min old-max new-min new-max)
  ;; scale the size of x by the proportion between ranges
  ;; and then shift into the new coordinate system.
  (let ((old-range (- old-max old-min))
        (new-range (- new-max new-min))
        (old-size (- x old-min)))
    (+ (* (/ new-range old-range)
          old-size)
       new-min)))
```

The **`rescale`** function takes a value *x* that lies within the inclusive boundaries *old-min* to *old-max* and returns a value equivalent to *x* that lies between the boundaries *new-min* to *new-max*. The values *x*, *old-min* and *old-max* are first converted to proportional quantities by subtracting out *old-min*. The local variable *old-range* is the difference of *old-max* and *old-min*, while *new-range* is the same for *new-max* to *new-min*. The body of the function is a single expression comprised of a division, a multiplication and an addition. Division and multiplication are inverse scaling operations and addition performs the offset. *old-range/new-range* is the ratio of the magnitudes of the two ranges. This ratio scales **`old-size`**, the magnitude of the value **`x`**. If *new-range* is larger than *old-range* then the multiplication will increase **`old-size`** by that proportion. If *new-range* is smaller than *old-range* then **`old-size`** will decrease in the same proportion. The last expression offsets the rescaled **`old-size`** by *new-min*, the minimum value of the new coordinate system, to shift the quantity back within the y boundaries specified to the function. A few sample interactions will suffice to show how the function operates:

Interaction 16-1. Examples of rescale.

```
cm> (rescale 5 0 10 10 20)
15
cm> (rescale 5 0 10 0 40)
20
cm> (rescale 5 0 10 100 200)
150
cm> (rescale 5 0 10 -200 0)
-100
cm> (rescale 2 0 10 200 100)
180
cm>
```

The first two interactions demonstrate offsetting and scaling in isolation. The first expression shifts, or offsets, the value 5, which lies half way between old boundaries 0 and 10, to the value 15, which lies half-way between the new values 10 and 20. The second interaction scales the value 5 to a value four times as large. The remaining three interactions combine scaling and offsetting. The third interaction rescales 5 to lie halfway between the boundaries 100 and 200. The fourth interaction transforms 5 to lie within negative boundaries. The last interaction inverts 5 within the new boundaries by reversing the *new-max* and *new-min* values.

## Points on a Curve To Find?

We now use the **`rescale`** function to define a musical process that maps values of the **`sin`** function, which range from -1 to 1, onto a range of MIDI key numbers.

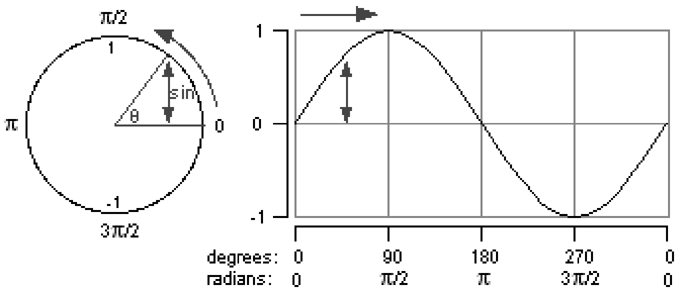
Example 16-2. Playing MIDI notes along a sine curve.

```
(define (play-sine len cycs key1 key2 rhy dur amp)
  (let ((maxx (- len 1))
        (maxr (* 2 pi cycs)))
    (process for x from 0 to maxx
             ;; rescale i to radians
```

```
          for r = (rescale x 0 maxx 0 maxr)
          ;; calc sine value -1 to 1 of r
          for a = (sin r)
          ;; rescale sin to keynum
          for k = (rescale a -1 1 key1 key2)
          output
          (new midi :time (now)
               :keynum k
               :amplitude amp
               :duration dur)
          wait rhy)))
```

**`play-sine`** plays MIDI events as discrete points along *cycs* cycles of a “sinewave”. We learned in Chapter 10 that a sinewave describes a smooth oscillation between a minimum and a maximum amplitude over time. Instantaneous points along this smoothly changing shape can be calculated using the trigonometric function **`sin`**. The **`sin`** function computes the ratio of the hypotenuse to the opposite side of an angle as it sweeps around the unit circle (Figure 1.) The angular input to **`sin`** is expressed in term of *radians*, not degrees. Radians are just another way to measure angles, and there are *2pi* radians in 360º.

Figure 16-1. The sine function.



As the angle sweeps from 0 to 2pi radians (0-360º) the value of **`sin`** produces one oscillation between -1 and 1. At 0 radians **`sin`** is 0, at pi/2 radians (90º) it is 1, at pi radians (180º) it is 0 again, and at 3pi/2 radians (270º) it is -1. The cycle repeats itself every 2pi radians (360º), which is same as 0.

**`Play-sine`** uses **`rescale`** to map a counter *i* onto an equivalent radian value *r* and to map values of **`sin`** onto key numbers. *Len* is the total number of events that **`play-sine`** creates, and *cycs* is the number of cycles of the "sine wave" produced in *len* events. Since there are 2*pi radians in one cycle of a sine wave, the maximum radian value *maxr* that the process will produce is *2*pi*cycs* radians over *cycs* cycles of the sine wave. Each time the process runs *r* is set to the current radian value determined by rescaling the *x* counter into radians. The current sine amplitude value *a* is then calculated and converted to the current key number *k* using the **`rescale`** function again.

Interaction 16-2. Listening to play-sine.

```
cm> (events (play-sine 80 4 20 100 .1 .1 .6)
            "sine.midi")
"sine-1.midi"
cm> (events (list (play-sine 80 4 20 100 .1 .1 .6)
                  (play-sine 60 5.7 50 80 .1 .1 .6))
            "sine.midi"
            '(0 2))
"sine-2.midi"
cm>
```

- sine-1.midi

- sine-2.midi

In the next example we define a variation of the **`play-sine`** process that calculates MIDI notes along the sine curve based on the current time of the process. This version allows us to compare different densities of events along the curve. We will also use the **`rescale`** function to control amplitude using the same sin values that control key numbers.

Example 16-3. Variation of sine wave process.

```
(define (play-sine2 end cycs rate keymin keymax ampmin ampmax)
  (process for r = (rescale (now) 0 end 0 (* 2 pi cycs))
           for a = (sin r)
           while (< (now) end)
           output (new midi :time (now)
                        :keynum (rescale a -1 1 keymin keymax)
                        ;; invert sine shape for amplitude
                        :amplitude (rescale a -1 1 ampmax ampmin))
           wait rate))
```

*End* is the stopping time for the algorithm and *rate* is the speed between MIDI events in seconds. *Cycs* is the total number of cycles to produce: as *cycs* increases for a given *rate* there will be fewer and fewer events per cycle of the sine curve.

Interaction 16-3. Listening to play-sine2.

```
cm> (events (play-sine2 10 .1 6 20 70 .1 .9) "sine.midi")
"sine-3.midi"
cm> (events (play-sine2 10 .5 6 20 70 .1 .9) "sine.midi")
"sine-4.midi"
cm>
```

- sine-3.midi
- sine-4.midi

## Linear Interpolation

The mathematical operation that **`rescale`** performs is called *linear interpolation*. The *x* on which the equivalence is based is called the independent variable and the output value that **`rescale`** calculates is called the dependent variable, or *y*. We can use a loop to collect pairs of *x* and *y* interpolation values into a list:

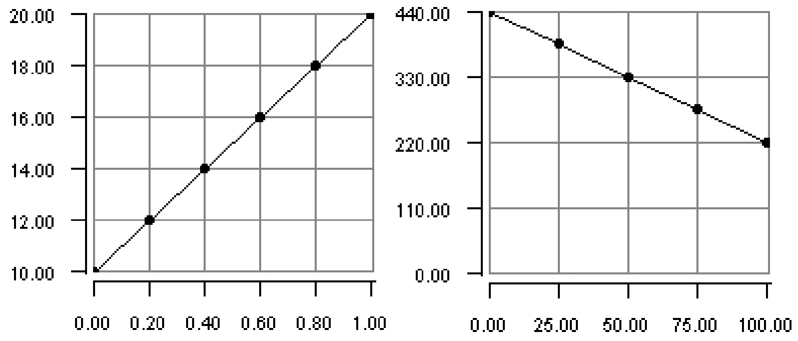
Interaction 16-4. X and y interpolation pairs.

```
cm> (loop for x in '(0 .2 .4 .6 .8 1)
          for y = (rescale x 0 1 100 200)
          collect x collect y)
(0 100 0.2 120.0 0.4 140.0 0.6 160.0 0.8 180.0 1 200)
cm> (loop for x in '(0 .25 .5 .75 1)
          for y = (rescale x 0 1 440 220)
          collect x collect y)
(0 440 0.25 385.0 0.5 330.0 0.75 275.0 1 220)
cm>
```

The linear relationship between these values can be shown by graphing the x and y pairs where the independent *x* value ranges along the horizontal, or *x* axis, and the dependent *y* values are range along the vertical, or *y* axis

(Figure 2).

Figure 16-2. Graphic display of x and y pairs in Example 2.



The points in the plot mark the equivalent positions of values along both axes. We can see in both graphs that linear interpolation produces values on a straight line between a minimum and maximum value and that the *slope* of the line may be upward or downward.

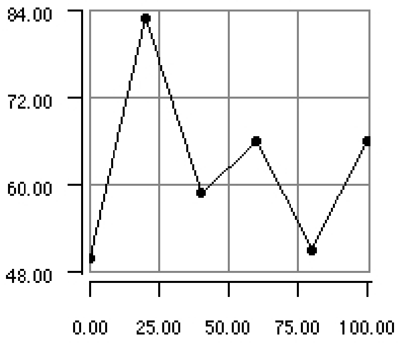
## Envelopes

The list of x and y pairs in Interaction 16-4 form a data structure called an *envelope*. Each pair of x y values are *coordinates* of points that can be plotted in a two dimensional plot such as those shown in Figure 2. There are only two requirements for defining envelope lists. First, an envelope must contain pairs of values and second, the *x* coordinates in the pairs must be in monotonically increasing order. The second requirement insures that only one unique *y* value exists for every unique *x* value. Although the relationship between x coordinate values is fixed, there is no ordering on y values, they may range anywhere within the minimum and maximum y axis values (Interaction 16-5 and Figure 16-3).

Interaction 16-5. x and y interpolation pairs.

```
cm> (loop for x in '(0 .2 .4 .6 .8 1)
          for y = (between 40 80)
          collect x collect y)
(0 50 20 83 40 59 60 66 80 51 100 66)
cm>
```

Figure 16-3. Graphic display of x and y pairs in Example 2.



Envelopes are a very simple but powerful data structure because each pair of points in an envelope can define the independent and dependent range specifications for linear interpolation. Common Music provides two functions that perform interpolation with envelopes:

[Function]

(**interp** *x x1 y2 x2 y2 ...*)

[Function]

(**interpl** *x env* [:scale *s*] [:offset *o*] [:base *b*])

Both functions perform linear interpolation over a series of point specifications but differ in the format of their function call arguments. The first argument to both functions is the independent *x* value to interpolate. The other arguments to **interp** are the sequences of *x* coordinate pairs that define the shape of the envelope. In contrast, the **interpl** function takes an envelope list as its second argument. The following two calls in Interaction 16-6 are equivalent:

Interaction 16-6. interp and interpl

```
cm> (interp .5 0 0 1 100)
50
cm> (interpl .5 '(0 0 1 100))
50
cm>
```

The **interpl** function also supports a set of optional *keyword arguments* after the envelope specification. Recall from Chapter 3 that keywords are symbols preceded by a colon. A keyword argument to a function is a special argument that is expressed as a pair: *:keyword value* where *:keyword* is the keyword name of the argument and *value* is its value.

Interaction 16-7. Passing keyword arguments to the interpl function.

```
cm> (interpl 0.5 '(0 0 1 1) :scale 100)
50.0
cm> (interpl 0.5 '(0 0 1 1) :offset 100)
100.5
cm> (interpl 0.5 '(0 0 1 1) :scale 100 :offset 100)
150
cm>
```

The next example demonstrates how **interp** can be used to control parameter values in a musical process. The process interpolates amplitude and tempo values using two **interp** expressions. Each interpolation divides the total length of the process into two segments on either side of a randomly chosen midpoint *mid* between 20 and 80 percent of the total length of the process. Each interpolation produces increasing values over the first segment and decreasing values over the second.

Example 16-4. Interpolating amplitude values from event number.

```
(define (play-interp len rate key1 key2 mintem maxtem minamp maxamp)
  (let* ((end (- len 1))
         (mid (+ (* .2 len) (random (* .8 len)))))
    (process for x from 0 to end
             for a = (interp x 0 minamp mid maxamp end minamp)
             for c = (interp x 0 mintem mid maxtem end mintem)
             output (new midi :time (now)
                         :keynum (+ key1 (random key2))
                         :amplitude a
                         :duration (* rate 1.5 c))
             wait (* rate c))))
```

In Interaction 16-8 we use a loop to sprout six versions of the process, each with a start time .5 seconds later than the preceding one.

Interaction 16-8. Listening to play-interpl

```
cm> (events (loop for k in (keynum '(c6 f5 bf4 ef4 af3 df3))
                  for s from 0 by .5
                  collect (play-interp 40 .2 k (+ k 7)
                                       1 .6 .4 .8))
            "interp.midi")
"interp-1.midi"
cm>
```

- interp-1.midi

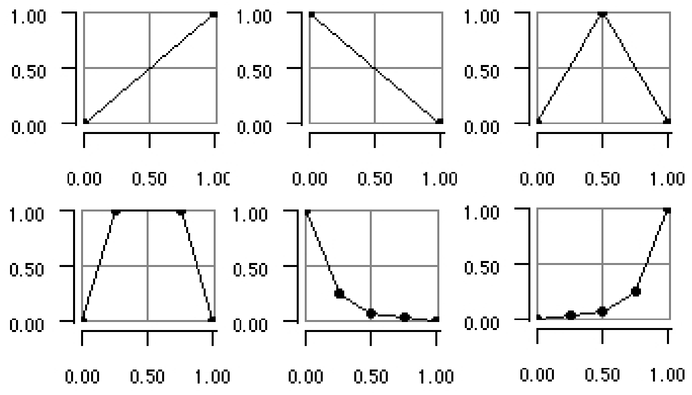
## Normalized Envelopes

Envelopes are a very simple yet powerful data structure in computer composition because they permit many different kinds of shapes to be defined and applied to parameterized data (Example 16-5 and Figure 16-4). A normalized envelope is an envelope whose *xy* point coordinates are defined within the range 0 to 1. Normalized envelopes express proportion, or shape, rather than direct values. Since shapes are generally useful tools many composers define global variables to hold the most common envelopes they work with. Example 16-5 shows some sample definitions to try (their shapes are shown in Figure 16-4):

Example 16-5. Common envelope shapes.

```
(define ramp-up '(0 0 1 1))
(define ramp-down '(1 0 0 1))
(define tri-env '(0 0 .5 1 1 0))
(define mid-env '(0 0 .25 1 .75 1 1 0))
(define exp-down '(0 1 .25 1/4 .5 1/16 .75 1/32 1 0))
(define exp-up '(0 0 .25 1/32 .5 1/16 .75 1/4 1 1))
```

Figure 16-4. Plots of envelope shapes.



To use a normalized envelope the *x* lookup rescales the *y* output value so that it lies within the appropriate range. The **`interpl`** function can perform all theses actions in a single function call.

Example 16-6. Applying normalized shapes to a process.

```
(define (shaper len env rate ampscale ampoff keyscale keyoff)
  (let ((last (- len 1)))
    (process for i below len
            for x = (/ i last)
            for a = (interpl x env :scale ampscale
                              :offset ampoff)
            for k = (interpl x env :scale keyscale
                              :offset keyoff)
            output (new midi :time (now)
                         :duration rate
                         :keynum k
                         :amplitude a)
            wait rate)))
```

Interaction 16-9. Calling shaper with different envelopes.

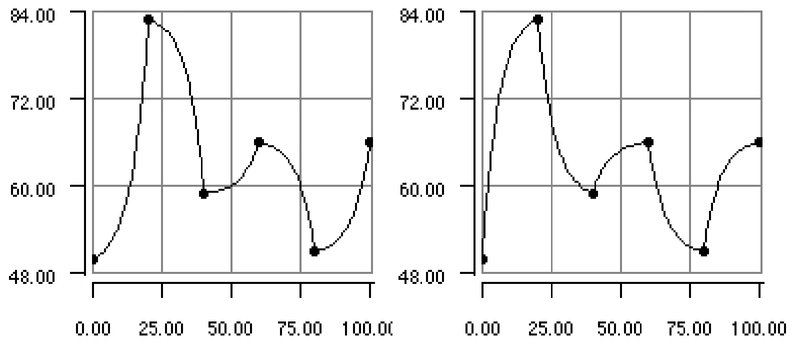
```
cm> (events (shaper 20 tri-env .15 .7 .05 16 48)
            "shaper.midi")
"shaper-1.midi"
cm> (events (shaper 20 exp-down .15 .5 .2 20 30)
            "shaper.midi")
"shaper-2.midi"
cm>
```

- shaper-1.midi
- shaper-2.midi

## Exponential Scaling

Linear interpolation is a wonderful tool, it provides a simple and general way to control sound parameters as a function of a dynamic process whose domain may not be directly related to the parameters in question. However, there are times when the straight line effect produced by linear interpolation may be inappropriate. For example, changes in tempo and amplitude often sound more natural if they are produced from *exponential scaling* rather than linear interpolation. Values generated by exponential scaling lie on a power curve rather than on a straight line. The *:base* keyword argument to **`interpl`** can be used to produce interpolation along a power curve rather than a straight line. If *:base* is 1 (the default) then linear interpolation results. But if *:base* is greater than 1 the interpolated values lie on a convex curve for positive slopes, or concave curves if it is less than 1 (Figure 16-5.) The more *:base* deviates from 1 the steeper the power curve between points.

Figure 16-5. Exponential interpolation using the envelope in Figure 3.



Common Music provides several additional functions for working with exponential scaling.

[Function]
(**`expl`** *x* `[:y0` *n*`] [:y1` *num*`] [:base` *num*`])`
[Function]
(**`explsegs`** *len sum base*)

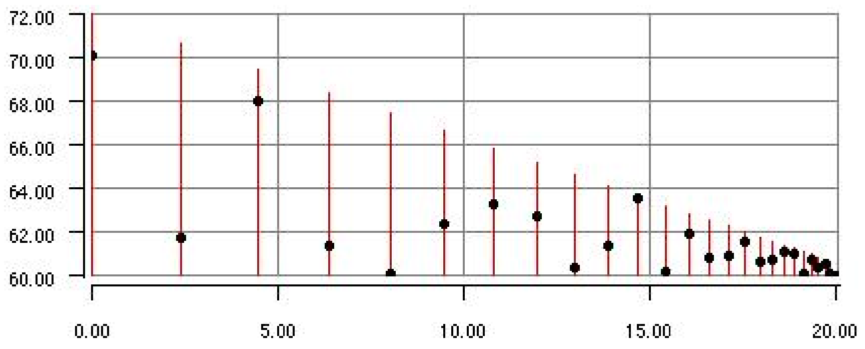
The **`expl`** function returns an exponential value between *:y0* and *:y1* for the given *x* value that must range from 0 to 1. **`Explsegs`** returns a list of *len* number of exponentially scaled segments that add up to *sum*. The *base* arguments to these functions are the same as for **`interpl`**. **`explsegs`** allows the composer to set the total size for some quantity and then chop it up into a number of exponentially related segments. The next example defines a process called **`ballfall`** whose tempo and key numbers are controlled by a list of exponential values produced by **`explsegs`**.

Example 16-7. Exponential curve for falling ball.

```
(define (ballfall drops end curve key amp)
  (let* ((all (explsegs drops end curve))
         (max (first all))
         (low (hertz key)))
    (process for d in all
             for e = (random (expt 2.0 (/ d max)))
             for k = (keynum (* low e) :hz)
             for a = (rescale d 0 max .1 amp)
             output (new midi time (now)
                         :duration d
                         :keynum k
                         :amplitude a)
             wait d)))
```

*Drops* are the number of times the ball bounces and *end* is the time it takes to decay. *curve* controls the steepness of the power curve. Since the longest drop occurs at the beginning the curve value should always be less than 1 for this process. *Key* is the lowest key number the ball will play and *amp* is the maximum amplitude. The variable *all* is set to the list of exponential values returned by **`explsegs`**. Since *curve* is less than 1 the first value in this list is the maximum drop value *max*. The process iterates for each drop value *d* in *all*. The variable *e* is set to a random value between 1 and 2 based on the proportion of the current drop *d* to the longest drop *max*. This means that as the process iterates the range for successive random values of *e* will diminish from 2.0 to a value approaching 1.0 (Figure 16-6.) This value is then used to transpose the *low* frequency somewhere within an octave and *k* is set to the key number of that Hertz frequency.

Figure 16-6. X axis plots start time, y axis plots range (red lines) and value of random key number (for *drop=25, end=20, curve=1/20).*



The amplitude variable *a* is determined by rescaling the current drop value to a value between .1 and the maximum amplitude specified to the process.

Interaction 16-10. Exponential curve for tempo and key number.

```
cm> (events (ballfall 40 10 1/256 60 .8) "ball.midi")
"ball-1.midi"
cm> (events (ballfall 100 15 1/150 40 .8) "ball.midi")
```

```
"ball-2.midi"
cm>
```

- ball-1.midi
- ball-2.midi

## Self Similar Scaling and Recursion

Scaling and offsetting can be combined to produce patterns that consist entirely of a single shape that is projected in identical proportions at smaller and smaller magnitudes. This kind of pattern is called a *self-similar*, or *fractal* pattern. Self-similarity can be found not just in visual patterns, but in musical ones as well. For example, the intervals in the chromatic scale are derived from a single ratio of $2^{1/12}$ so that intervals in a melody sound the same when transposed to different octaves even though the actual Hertz frequency values between the tones are not equivalent. A self-similar pattern can be created by applying the same set of rules (i.e., an algorithm) over and over again at different levels of magnitude until enough of the pattern has been generated to suffice.
Self-similarity is easiest to understand when applied to a single dimension and most naturally implemented using a *recursive* function definition. A recursive function is a function that calls itself to solve successively simplified versions of a problem. In order to illustrate self-similarity and recursion we will create a function that simply "chops" a value up into self-similar segments at smaller and smaller magnitudes and then returns a list of all the segments that it calculates. For example if an input of 8 were chopped into 2 segments over 4 levels of magnitude the algorithm would return the list:

(8 4 2 1 1 2 1 1 4 2 1 1 2 1 1)

which corresponds to the levels and proportions in Table 1

Table 16-1. Levels and proportions of a self-similar organization.

| | | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| **Level 1** | 8 | | | | | | | |
| **Level 2** | 4 | | | | 4 | | | |
| **Level 3** | 2 | | 2 | | 2 | | 2 | |
| **Level 4** | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 |

The **sschop** function Example 16-8 is a recursive algorithm that calls itself to reproduce the exact same behavior at different levels of magnitude. Its recursive nature means that **sschop** reflects self-similarity in its definition as well as in the results it returns.

Example 16-8. Self-similar segments of a line.

```
(define (sschop value segments levels)
  (if (< levels 1)
    '()
    (let* ((nextval (/ value segments))
           (nextlev (- levels 1))
           (chopped (loop repeat segments
                          append (sschop nextval segments nextlev))))
      (list* value chopped))))
```

The function **sschop** takes three inputs: a *value* to chop, the number of *segments* to chop *value* into, and the number of magnitude *levels* to chop over. The function returns a list of chopped values. The body of **sschop** consists of a single **if** expression. If the value of *levels* is less than 1 then **sschop** returns an empty list (Interaction 16-11). If *levels* is greater than zero then **sschop** calculates a list of values to return. These values consist of the current value added to the front of a list containing successive levels of chopped values at proportionally smaller sizes. The **`let*`** establishes three variables. *nextval* is the proportionally smaller next value, computed by dividing the current *value* by the number of segments in each level. *nextlev* is set to *level-1*. *Chopped* holds the list of segments that constitute all the values under *nextlev*. This value is computed by a loop that iterates *segments* number of times and appends the results of a recursive call to *sschop*w at the next smaller size and with *levels* reduced by one. Since *sschop* always returns a list, the **`append`** clause is used to concatenate each sublist into one large list. The **`list*`** function then adds *value* to the front of this list so that the current level's value is in front of all of its segments. Interaction 16-11 shows several sample calls to **`sschop`** that demonstrate its self-similar nature.

Interaction 16-11. Calling sschop.

```
cm> (sschop 440 2 0)
()
cm> (sschop 440 2 3)
(440 220 110 110 220 110 110)
cm> (sschop 36 3 3)
(36 12 4 4 4 12 4 4 4 12 4 4 4)
cm>
```

## Sierpinski's Triangle

Sierpinksi's Triangle is a famous two dimensional fractal. The rules to create Sierpinski's triangle are simple:

1. Start with a triangle.
2. Draw a smaller triangle in the middle of the larger triangle with the points of the smaller triangle on the midpoints of the larger triangle's sides. This partitions the large triangle into 4 triangles: the middle triangle and the triangles at each of the corners of the larger triangle.
3. Leave the middle triangle empty.
4. Visit each of the corner triangles in the larger triangle and repeat the process using each of these as the starting triangle.
5. Repeat for as many levels as you want: the result is Sierpinski's Triangle.

We will now transform our **`sschop`** algorithm into a musical process that generates a self-similar melody as an "homage" to Sierpinski's Triangle. The process definition and its output were already seen in Chapter 10. The algorithm is presented here again followed by its explanation. Rather than concerning ourselves with two dimensional mapping (see Chapter 17) our algorithm will use the basic method described in **`sshop`** to generate its self-similar melody out of set of tones, where each tone represents the side of a triangle.

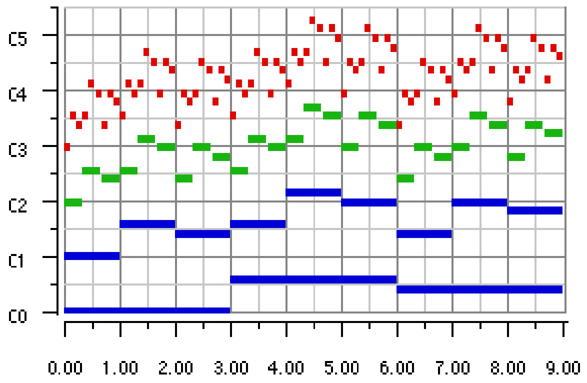
Example 16-9. Sierpinski's Melody

```
(define (sierpinski tone melody levels dur amp )
  (let ((len (length melody)))
    (process for i in melody
             for k = (transpose tone i)
             ;; play current tone in melody
             output (new midi :time (now) :duration dur
                         :amplitude amp :keynum k
                         :channel 0)
             when (> levels 1)
             ;; sprout melody on tone at next level
             sprout (sierpinski (transpose k 12)
                                melody
                                (- levels 1)
                                (/ dur len)
```

```
                                       amp)
             wait dur)))
```

The **sierpinski** process reproduces a melodic shape on successive degrees and octaves of itself. *tone* is the current melodic tone to play, *melody* is a list of intervals representing the melody, *level* is the number of levels of the melody to produce, and *dur* and *amp* are the tone's duration and amplitude. The variable **len** is set to the number of intervals in the *melody*. The process first plays the current *tone* in a channel for each level. If *levels* is greater then 1 then the process sprouts a recursive definition of itself to generate the next level. The next level's tone is calculated by transposing the current tone in the melody up one octave. The duration for each tone in the next level is *dur/len*, the current level's duration divided by the number of intervals in the melody. Thus, the entire melody in the next level will occupy the same amount of time as the current tone on which it is based (Figure 16-7).

Figure 16-7. Self-similar melody from sierpinski.



The value for *levels* is decremented by 1 , which will cause the recursive generation to finally stop when this value reaches 0.

Interaction 16-12. Listening to sierpinski.

```
cm> (events (sierpinski 'a0 '(0 7 5 ) 4 3 .5) "sier.midi")
"sier-1.midi"
cm> (events (sierpinski 'a0 '(0 -1 2 13) 5 24 .5) "sier.midi")
"sier-2.midi"
cm>
```

- sier-1.midi
- sier-2.midi

Specify levels and melody with care! The number of events the **sierpinski** process generates is exponentially related to the length *m* of the melody and the number of levels *l*:

$$m \cdot \sum_{i=0}^{l-1} m^i$$

Accordingly, the first events call in Interaction 16-12 generates 120 events and the second produces 1364.

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file mapping.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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H. Taube
06 May 2003

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# 17 *Etudes*, Op. 6: Sonification of Chaos

Thru all the tumult and the strife,
I hear that music ringing.
It sounds and echoes in my soul;
How can I keep from singing?

— Quaker, Worhsip Comes Alive

*Sonification* means to use an acoustic signal to perceive information inherent in a collection of data (see the on-line report Sonification Report: Status of the Field and Research Agenda by Gregory Kramer for an overview of research in this area.) Sonification is to the ear what graphic plotting is to the eye — a way to understand relationships that would otherwise be difficult or impossible to perceive. Our ears are very keen instruments that can detect subtle, multidimensional changes in sound space. We are probably all familiar with listening to minute variations in sound in order to detect the source of a mechanical problem. Sonification can yield information about regularity, overall shape, grouping, transitions, and deviations in the data that is presented as an acoustic signal. *Parameter mapping* is one of the most common ways to implement sonification. By projecting data onto frequency, amplitude and timbre over time many dimensions of data can be perceived and related simultaneously. In this Etude we apply the mapping techniques presented in Chapter 16 to data sets generated by several chaotic functions. The text and examples in this Etude were written by Tobias Kunze for a workshop on algorithmic composition held at CCRMA, Stanford University. Our aims in the Etude are twofold: to understand how mapping can be implemented given a set of non-musical equations, and to demonstrate how sonification can reveal the behavior of non-musical functions. The term *deterministic chaos* refers to the behavior of systems that are not random but are nevertheless unpredictable in nature. The physical world is filled complex dynamic processes like turbulence and weather that are deterministic but whose behavior is nevertheless impossible to predict. There are several features that characterize these systems. First, chaotic systems are highly sensitive to initial conditions — in one initial arrangement their elements produces periodicity; in another, closely proximate arrangement, complete unpredictability ensues. This is because chaotic system are *nonlinear*, which means that changes in one element do not produce a like, or equivalent change in others. Because of nonlinearity, cause and effect relationships are not proportional in a chaotic system, so that two functions for a single equation may evolve in a similar way but then suddenly diverge into different paths (Figure 17-3). The *butterfly effect* observed first by the meteorologist Edward Lorenz encapsulates the sensitivity and nonlinear nature of these systems quite well: A butterfly flaps its wings in Tokyo and it rains in New York.

## Op. 6 No. 1, The Logistic Map

The *Logistic Map* is a simple example of a discrete dynamical system that actually names a whole family of iterative functions described by the Logistic Equation:

$$f_{n+1} = cf_n(1-f_n); \quad \text{for } 0.0 \le c \le 4.0$$

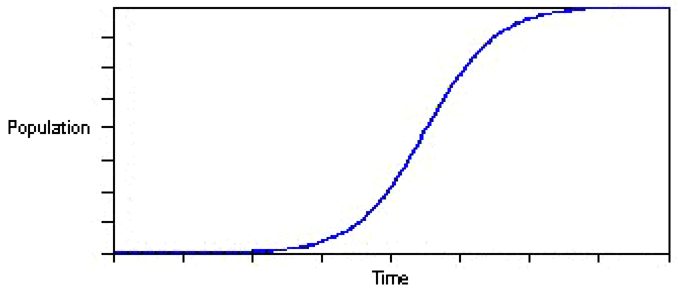
Each real value of $c$ in the equation yields a different dynamic behavior and hence a different function.

### Background

The Logistic Map models population growth in a basic *activator-inhibitor* system. Although population growth rates are constant in an ideal environment, actual growth rates are not due to limited resources such as food and water and natural catastrophy. If resources are limited, the growth rate slows down as the population grows and eventually reaches a *limit* or *attractor* at which point the population is at maximum capacity for the environment.

The S-shaped curve of population growth in such an environment is known as the *logistic curve* (Figure 17-1).

Figure 17-1. Expected growth curve.

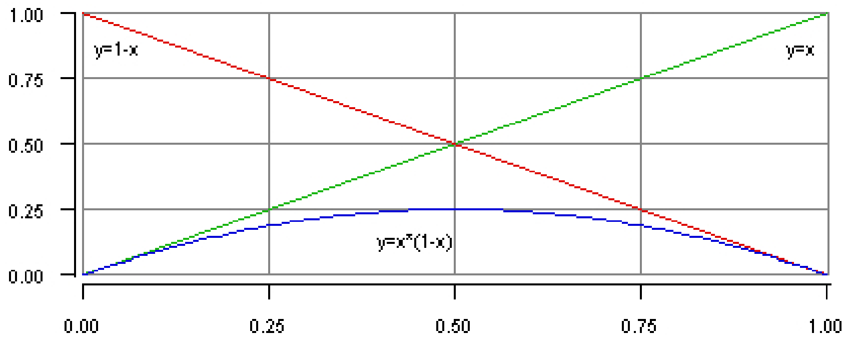


Resource limitations may be viewed as "passive" inhibitors, that is, they only slow down population growth. However, most environments also involve "active" inhibitors, such as epidemics, which reduce a population or at times cause extinction. Whether populations in such an environment approach a maximum, die out or tend to converge towards a number in between these extremes is hard to predict, since most systems with active inhibitors behave in a chaotic manner. Each function of the *logistic map* may serve as an algebraic model of such a chaotic environment.

## Activation and Inhibition

As a model for the development of populations (and similar processes), the logistic map may be broken down into an activating and inhibiting term, related by a function *R* and with a weighting constant *c* applied to one of the terms: $f=(c * activator) R Inhibitor$. Now, *R* happens to be the multiplication operator in the logistic map (Figure 17-2) although other possibilities are imaginable.

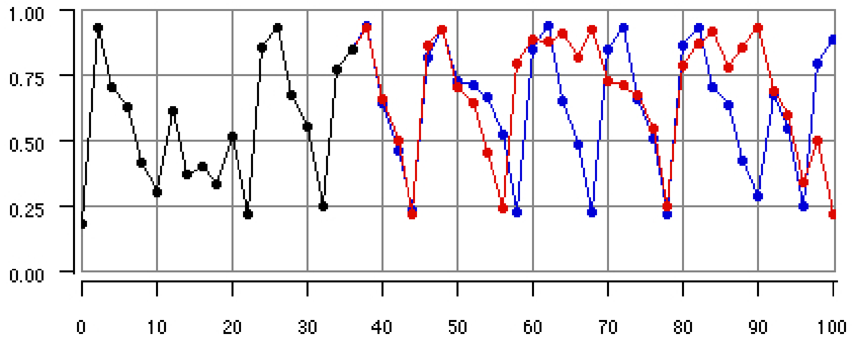
Figure 17-2. Activator term *x* (green) and inhibitor term *1-x* (red) of the logistic map and their product (blue), which exhibits a global maximum of 0.25. To keep the final product $c*x(1-x)$ stable between 0.0 and 1.0 it is required that $0 \leq c \leq 4.0$.



## Chaos and The Butterfly Effect

In contrast to the many random processes which produce *indeterministic* behavior, the logistic map is strictly deterministic. For instance we can iterate a given function of the logistic map *n* times to determine its *nth* value. The *nth* value of the function is determined solely by the initial value, $f_0$and the function constant *c*. But exactly how the *nth* value depends on the initial conditions is hard to predict. Figure 17-3 parallels two processes with only minimally altered initial values. This difference leads after an initial phase of almost parallel evolution to totally different behavior in both processes. It is almost impossible to predict the exact iteration at which the two processes take on different behavior.

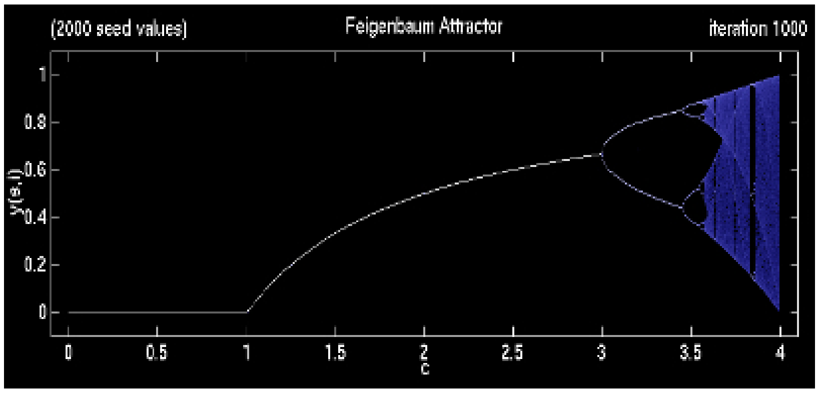
Figure 17-3. Development of the logistic function for $f_0$=.95 and $f_0$=.949999999 (blue) for $c$=3.75 in both cases. This minimal variation of the initial value yields dramatically different sequences for $n>30$.



## Bifurcation

The behavior of the logistic map does not depend on the value of $f_0$ alone but also on the constant *c*. In fact, for numerous values of *c*, *f*0 doesn't matter much at all since the function will converge eventually. Most obviously, setting $c$=0 yields zero values for all values of $f_n$ for $n$>0. (Figure 17-4) Similar behavior results for all values of $c$<~3.0. At this point, the map converges to several attractors again, depending on the value of the initial $f_0$. It converges first to 2, then to 4, 8, 16 ... attractors until it reaches chaotic behavior. (This repeated doubling within families of functions such as the logistic map has been first noted and extensively studied by the physicist Mitchell J. Feigenbaum.) Interestingly, the sequence of functions of the logistic map traverse several states of temporary tranquility on their way to sheer chaotic behavior . Thus, $f_n$ is actually a family of functions of the form $y(s,c) = c * y * (1 - y)$, with seed values $s$ = {0.0005, 0.001 ... 1.0}. Although the logistic map may give radically different numerical sequences for any two functions of the same family (functions with equal $c$ but differing seed value $s$) the distribution pattern of a family after $i$ iterations and the histogram of any of its functions up to the $i$th iteration approach the same attractor.

Figure 17-4. Bifurcation of the logistic map. Values for $c$ are plotted on the $x$-axis range from 0.0 to 4.0 and are mapped onto the distribution of values of $f(n)$ after 1000 iterations starting from 2000 equally distributed initial seed values of $f_n$. Ranging from black over blue to white, colors represent statistical density.



## Mapping the Logistic Map

We can now sonify the Logistic Map by mapping iterative values onto sound parameters.

Example 17-1. The Logistic Map as key numbers

```
(define (logmap chaos len rate dur key1 key2 amp)
  (let ((y (random 1.0))
        (k 0))
    (process repeat len
             set y = (* y chaos (- 1 y))
             set k = (rescale y 0 1 key1 key2)
             output (new midi time (now)
                         :keynum k
                         :duration dur
                         :amplitude amp)
             wait rate)))
```

The **`logmap`** process maps values from the Logistic Map onto key numbers between *key1* and *key2*. The initial conditions $f_0$ are represented by the specified *chaos* parameter, which may range between 0 less than 4.0, and the initial random value for *y*.

Interaction 17-1. Listening to the Logistic Map

```
cm> (events (logmap 3.7 200 .125 .25 60 96 .6) "logmap.midi")
"logmap-1.midi"
cm> (events (logmap 3.99 200 .125 .25 60 96 .6) "logmap.midi")
"logmap-2.midi"
cm>
```

- logmap-1.midi
- logmap-2.midi

The second example increases the *chaos* factor close to the maximum. In the next example we explore chaotic rhythms.

Example 17-2. Chaotic rhythms

```
(define (groove chaos len pulse)
  (let ((y (random 1.0)))
    (process repeat len
             set y = (* y chaos (- 1 y))
             output (new midi :time (now)
                         :keynum 60
                         :duration .01)
             wait (* pulse y))))
```

Interaction 17-2. Listening to the Logistic Map

```
cm> (events (groove 3.99 50 .5) "logmap.midi")
"logmap-3.midi"
cm>
```

- logmap-3.midi

This last example links rhythms and pitches together to form more distinctive gestures. The rhythms are quantized to increments of .1 using Common Music's **`quantize`** function.

Example 17-3. Two dimensional Logistic Map

```
(define (logmap-2d chaos len pulse key1 key2 dur)
  (let ((y (random 1.0)))
    (process repeat len
             set y = (* y chaos (- 1 y))
             output (new midi :time (now)
                          :keynum (rescale y 0 1 key1 key2)
                          :duration dur)
             wait (quantize (rescale y 0 1 0 pulse) .1))))
```

Interaction 17-3. Listening to logmap-2d

```
cm> (events (logmap-2d 3.99 200 .125 60 96 1) "logmap.midi")
"logmap-4.midi"
cm>
```

- logmap-4.midi

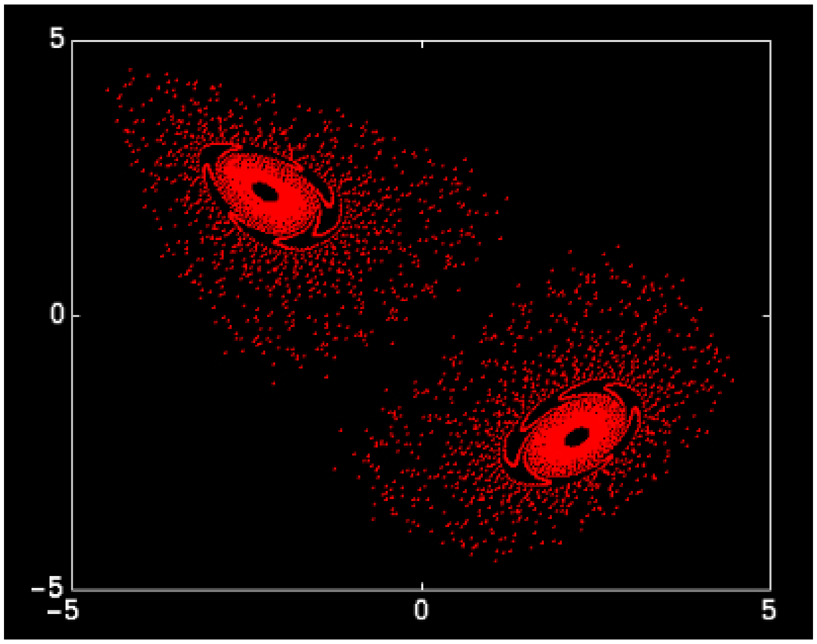
## Op. 6 No. 2, The Henon Map

The *Henon Map* is given by the two-dimensional function:

$$H(x,y) = ((y + 1) - ax^2, bx)$$

where $a$ and $b$ are constants, typically 1.4 and 0.3, respectively. The Henon Map is the two-dimensional analogue to the logistic function in that it has an oscillating catalytic term $y+1=bx+1$ and an inhibiting part $-ax^2$ (Figure 17-5).

Figure 17-5. The Henon Map for standard values *a*=1.4 and *b*=.3 after 5000 iterations.



We can define two processes to map the Henon into musical space. The first maps Henon values onto frequency, the second does this in two voices.

Example 17-4. Henon Mapping

```
(define (henon1 len rate dur key1 key2)
  (let ((x 0)
        (y 0))
    (process repeat len
             for z =  (- (+ y 1) (* .2 x x))
             set y = (* .99 x)
             output (new midi :time (now)
                         :keynum (rescale y -5 5 key1 key2)
                         :duration dur)
             wait (rescale z -5 5 0 rate)
             set x = z)))

(define (henon2 len rate key1 key2)
  (let ((x 0)
        (y 0)
        (k 0)
        (r 0))
    (process repeat len
             for z = (- (+ y 1) (* .2 x x))
             set y = (* .99 x)
             set k = (rescale y -5 5 key1 key2)
             set r = (rescale z -5 5 0 .4)
             output (new midi :time (now)
                         :keynum k
                         :duration r)
             sprout (new midi :time (+ (now) rate)
                         :keynum k
                         :duration .05)
             wait (+ rate r)
             set x = z)))
```

Interaction 17-4. Listening to Henon Mapping

```
cm> (events (henon1 250 .24 1 48 84) "henon.midi")
"henon-1.midi"
cm> (events (henon2 250 .25 48 84) "henon.midi")
"henon-2.midi"
cm>
```

- henon-1.midi
- henon-2.midi

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file chaos.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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06 May 2003

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# 18 Randomness and Chance Composition

There is divinity in odd numbers, either in nativity, chance, or death.

— William Shakespeare

Music composition, and artistic expression in general, often reflects a tension between rational and irrational processes that underlie and motivate creative work. Perhaps more than at any other time in history, the 20th century witnessed the rational and irrational taking explicit shape in the techniques that composers used to create music. It should not be surprising that the same century that gave birth to Quantum Mechanics and Heisenberg's Uncertainty Principle also found artists consciously incorporating *non-deterministic*, or random processes into their artistic pallettes. One of the most important expressions of this influence was in the *chance procedures* adopted by some artists. Whether it is motion added to a dance by Merce Cunningham, an abstract painting by Jackson Pollock or the I Ching music of John Cage, randomness and chance are consciously embraced as generative techniques for making art. An interest in chance procedures served as an important motivation in the development of the experimental music tradition in the 1950's, from which computer composition stems. The earliest examples of computer composition by Lejaren Hiller are largely based on chance procedures that will be discussed in the next two chapters. One reason that many composers who work with computers are attracted to chance operations is that these techniques can have a profound effect on the compositional process yet are so simple and easy to perform with a computer. The computer also allows composers to experiment with many different types of randomness with characteristics quite different than the simple coin tossing strategy that all composers are familiar with. It perhaps somewhat ironic then, that the computer programs that composers use to create chance compositions are not really generating randomness at all!

## Random Number Generators

Computers are designed to function in a totally consistent and reproducible manner. This is a wonderful feature when your plane is landing or your tax returns are prepared but it also means that random outcomes are inherently difficult to achieve with such a deterministic machine. Although some applications use specialized hardware to generate truly random signals, more typical computer programs only simulate randomness using a special type of algorithm called a *pseudo-random* number generator. Random number generators are software programs that calculate a series of numbers with statistically random properties. But as the name suggests, these numbers are not really random at all. A pseudo-random generator is a completely deterministic algorithm that generates one series of numbers. A common design characteristic in many random generators is to use each past output value as the *seed* for calculating the next number in the series. This means that each time the generator starts from a particular seed the exact same sequence of numbers in the series will ensue. Many random number generators, including Lisp's **`random`** function, allow the user to specify an initial random state seed that positions the algorithm to a certain location in its sequence of numbers. Composers who want to express randomness "philosophically" in a composition will want to use different initial seeds each time a chance program runs to insure the program generates a different outcome each time. On the other hand, composers also take the opposite approach and save seeds in order to "back up" in the event that a random process has produced particularly satisfying results! This is one way to implement the common *generate and test* compositional style in chance composition. By building random choice into the algorithms that generate music the composer can generate many possible "drafts" of a piece, each with its own random variations. These versions can then compared to determine which one(s) best satisfies the composer's goals. Sometimes a whole piece is generated in this manner, other times the composer collects the best features from different drafts and combine them into a final version. Since the initial seed for each draft was saved the composer can regenerate the best versions at any time, as long as the program code remains exactly the same.

Since the material in this book assumes that the reader is interactively experimenting with examples we are virtually assured that the same path through Lisp's random number series will not be repeated. For this reason none of the examples in this book make use of random number seeds. But by establishing different seeds each time you begin to work, or by saving the current seed before you run a chance program, you can minimize (or maximize) the likelihood that your program will generate the same results. To learn more about the process of seeding in Lisp consult the documentation on the **`random`** function in the Language specification.

## Probability

A random process is a process in which outcomes have only a certain likelihood of happening. *Probability* is a measurement of the likelihood of these events actually occurring. Probability in our daily lives is often described in terms of a "percentage chance". For example, the weatherman might say: "There is a 25% chance that it will rain today". The percentage indicates that probability is *distributed* over all the events in a random process, where each event, or range of events, has a certain proportion of the total probability. So, although we do not know for certain if it will rain or not, probability allows us to infer several facts from the forecaster's statement:

1. there is a 25% chance that it will rain today
2. there is a 75% chance that it will not rain today
3. there is a 100% chance that it will either rain or not rain today
4. there is a 0% chance that neither will happen

Weather forecasters notwithstanding, probability factors are typically expressed as values between zero and one rather than as percentages between 0 and 100. The probabilities of all the events in a random process add up to 1.0 (100%) because there must be absolute certainty (1.0) that at least one of the outcomes will happen.

Some random processes are characterized by an equal distribution of probability over all of the events in the process. This is called a *uniform probability distribution*. For example, when a fair die is rolled there is an equal likelihood that any one of the six sides will land face up. Since the total probability must add up to 1 and there is an equal chance for any side to land on top, each of the six sides has a probability factor of 1/6. For a given event $x$, its probability, P($x$), is the ratio of the number of ways the outcome can happen to the total number of possible outcomes. If $x$ is a randomly chosen a key on a piano then P($x$)= 1/88. If $x$ is a randomly chosen black key on the piano, then there are 36 possible ways to do this and so P($x$)=36/88=9/22. Similarly, the complementary probability of an event (all the events that are not $x$) is 1-P(x). So the probability of selecting a white key $x$ on the piano can be calculated by complementing the probability of black keys, or P(x)= 1-9/22=13/22. In a continuous distribution, probability is defined over an area under a distribution curve. This means that the probability of any single event, or "point" is zero since it has no area. This sounds odd but since probability is the ratio of the number of ways an event can happen to the total number of events then if there are an infinite number of events the probability of any single event is zero.

Lisp's **`random`** function generates numbers in both discrete and continuous distributions. By specifying an integer range to **`random`** only integers will be returned. If the range is expressed as a real, or floating point value, then continuous values within the limits of floating point precision are returned. The **`random`** function always returns a value between zero less than the bounds specified to the function. To generate any other range of values, or to randomly generate non-numeric values, requires that the output from **`random`** be scaled and offset, or used as indices into lists of non-numeric data. These kinds of manipulations are such a common occurrence in composition that Common Music provides a number of different functions to facilitate chance operations selection in musical contexts. These functions are listed together here and then discussed at appropriate places in the chapter.

[Function]

(**between** *low high* [*skip*])
[Function]
(**odds** *prob* [*then*] [*else*])
[Function]
(**pick** *...*)
[Function]
(**pickl** *list* [:start *i*] [:end *i*] [:skip *x*])
[Function]
(**ran** [:type *k*] [:a *n*] [:b *n*])
[Function]
(**shuffle** *list* [:start *i*] [:end *i*])
[Function]
(**tendency** *n env1 env2* [:offset *n*] [:scale *n*])
[Function]
(**vary** *n pct* [*where*])

## Probability Distributions

The probability of events in a random process is determined by a *probability distribution* — a mathematical function that describes the likelihood of all possible outcomes. Distributions can be characterized by a *mean* or average, value that they produce and by a characteristic "shape" that the probability expresses over the events in the random process. One of the great advantages to using a computer to work with random processes is that it is possible for a composer to employ complex distributions without needing to understand the underlying mathematics of their implementation (which can be quite complex for some types of randomness). Composers who are interested in a formal presentation of random number generation should read [Knuth], a seminal discuss that contains the definitions of many random number generators with different probability characteristics. [Dodge/Jerse] also contains an excellent discussion of basic probability theory and presents sample code. The source code to the *ran* function in Common Music can be studied to see how common non-uniform distributions are generated from the uniform **random** function in Lisp.

In the next few sections we define the characteristics of some common distributions. Each example includes a graphical plot of sampled events together with a *histogram* that shows a characteristic density of events that each probability distribution generates. The histograms were generated by analyzing the output from a musical process that maps random values onto a range of integer MIDI key numbers. The musical processes that generated the results are shown in each example. The musical process can be listened to in discrete mode — quantized to the closest MIDI key number — or in continuous mode, by using the channel tuning methods discussed in Chapter 14 to output microtonal frequency values. Example 18-1 contains the basic process definition we will use to sonify the distributions.

Example 18-1. Mapping distributions onto frequency space.

```
(define (play-ran type len rate key1 key2)
  (process repeat len
           for r = (ran :type type)
           for k = (rescale r 0.0 1.0 key1 key2)
           output (new midi :time (now)
                       :keynum k
                       :duration (* rate 1.5)
                       :amplitude .5 )
           wait rate))

;;; channel tuning on first 9 channels
```

```
(define ct9 (list true 0 8))

(define (mean-ran type n a b)
  (/ (loop repeat n
           sum (ran :type type :a a :b b))
     n))
```

The sound example provided for each distribution plays the distribution in continuous mode. The **`play-ran`** function maps values from Common Music's **`ran`** function to a range of key numbers specified to the process. The **`ran`** function generates random numbers in a distribution specified using the **`:type`** argument. The following distribution types are supported:

```
:uniform :low :high :triangular :beta :exponential :cauchy
:poisson :gamma :gaussian
```

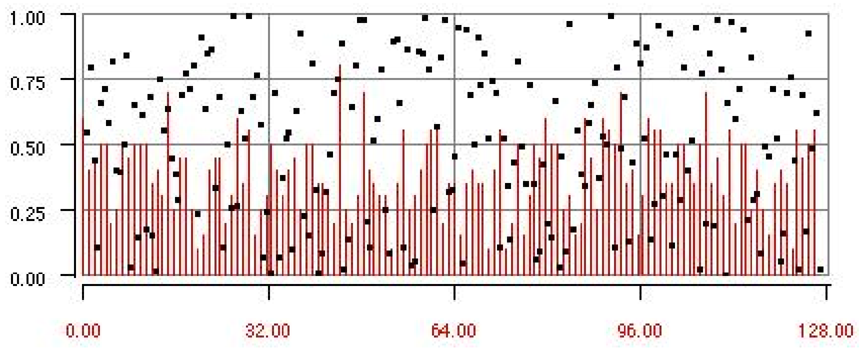
If the **`:type`** argument is not specified then values in uniform distribution are generated. See the documentation on **`ran`** in the Common Music Dictionary for more information about this function.

The **`mean-ran`** function calculates the *mean*, or average, value of *n* outcomes of **`ran`**. This gives us some measure of how well the pseudo-random number generator approximates the theoretical mean value of each distribution type.

## Uniform distribution

In a *uniform distribution* all events have an equal likelihood of being selected. This means that in a discrete selection over *n* events, each event has *1/n* probability of being selected. In a continuous distribution the events under two equal portions of the total probability range will have an equal likelihood of selection with a mean value of .5. Lisp's **`random`** function return numbers in a uniform distribution. These values can then be filtered or processed in different ways to produce non-uniform behavior.

Figure 18-1. Two plots depicting values (black points) and histogram (red bars) of uniform distribution. The Y axis plots continuous random values normalized 0-1. The x axis plots the histogram range defined over 127 key numbers.



Interaction 18-1. Playing continuous uniform distribution.

```
cm> (events (play-ran :uniform 100 .1 20 100) "uniform.midi"
            :channel-tuning ct9)
"uniform-1.midi"
cm>
```

- uniform-1.midi

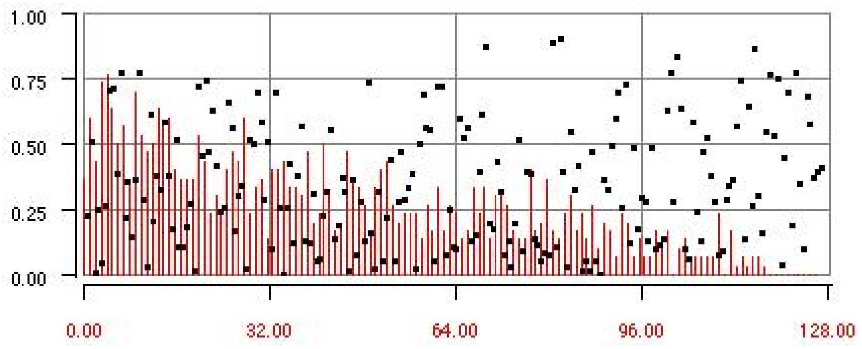
## Linear "Low-pass" Distribution

In a linear “low-pass” distribution the probability of larger numbers decreases linearly over the range of values returned in the process. The low-pass behavior is implemented by filtering the a uniform process so that the maximum of two uniform random numbers is returned:

```
        (min (random 1.0) (random 1.0))
```

The expected mean value of the low distribution is .2929.

Figure 18-2. Low pass values (black points) and histogram (red bars).



Interaction 18-2. Playing the low pass distribution.

```
cm> (events (play-ran :low 50 .1 20 100) "low.midi"
            :channel-tuning ct9)
"low-1.midi"
cm>
```

- low-1.midi

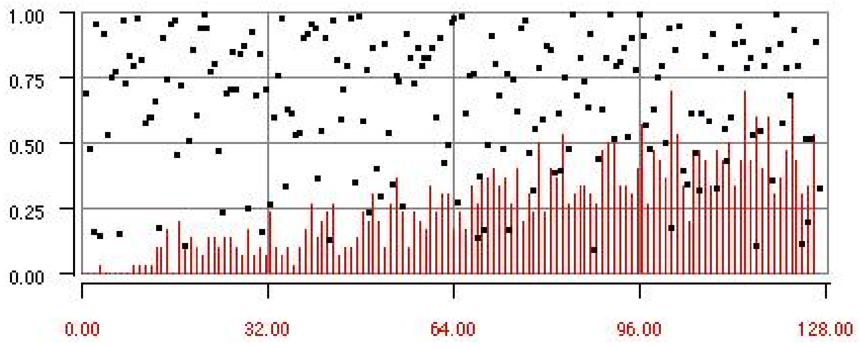
## Linear "High-pass" Distribution

In a linear “high-pass” distribution the probability of larger numbers increases linearly over the range of values. The distribution can be implemented by returning the maximum of two uniform random numbers:

```
        (max (random 1.0) (random 1.0))
```

The expected mean value of the high distribution is .2929.

Figure 18-3. High pass values (black points) and histogram (red bars).



Interaction 18-3. Playing the high pass distribution.

```
cm> (events (play-ran :high 50 .1 20 100) "high.midi"
            :channel-tuning ct9)
"high-1.midi"
cm>
```

- high-1.midi

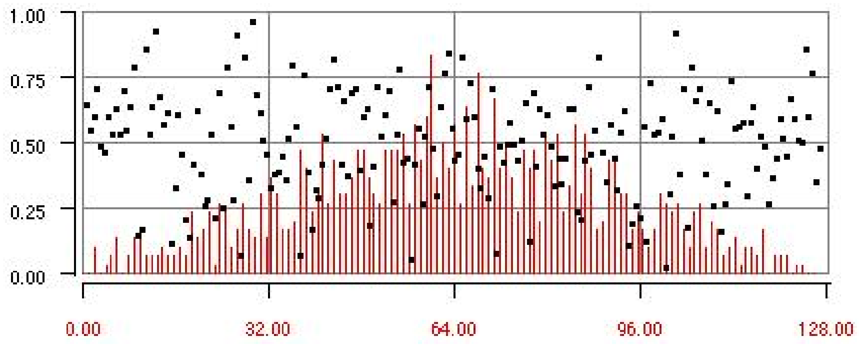
## Triangular Distribution

In a triangular distribution the probability of numbers in the middle of the distribution increases linearly. A mean distribution can be implemented by averaging two uniform random numbers:

```
(/ (+ (random 1.0) (random 1.0)) 2))
```

The expected mean value of the triangular distribution is .5.

Figure 18-4. Triangular distribution values (black points) and histogram (red bars).



Interaction 18-4. Playing the triangular distribution.

```
cm> (events (play-ran :triangular 50 .1 20 100) "tri.midi"
            :channel-tuning ct9)
"tri-1.midi"
cm>
```

- tri-1.midi

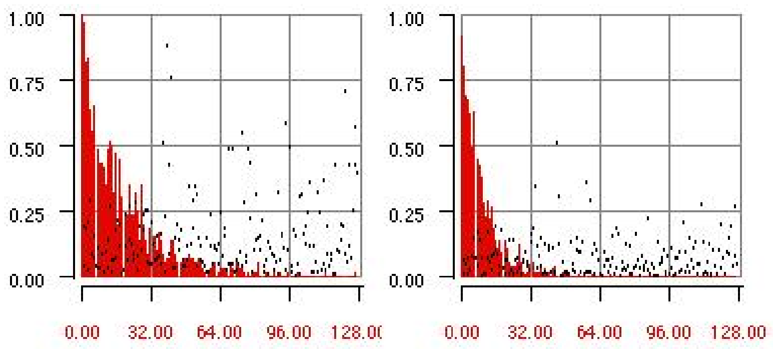
## Exponential Distribution

The exponential distribution returns the natural logarithm of a uniformly chosen number.

```
(/ (- (log (- 1 (random 1.0)))) a))
```

The *a* parameter provides a “stretching factor” that scales values closer or further from 0; the larger the value of *a* the more the values are pushed toward zero. If random returns 0 then $\log_e 1$ is 0, the minimum value in the distribution. As random values exceed *1/e* (0.367) the values returned in the distribution exceed 1.0. There is no upper limit to values produced in the exponential distribution, but 99% of the time its value is less than 6.908.

Figure 18-5. Exponential distribution values (black points) and histogram (red bars) for *a=1* and *a=2*.



The expected mean value of the exponential distribution for *a=1* is .693.

Example 18-2. An exponential distribution.

```
(define (play-exp len rate key1 key2 a)
  (process repeat len
           for r = (ran :type :exponential :a a)
           for k = (rescale r 0.0 6.908 key1 key2)
           when (< k 128)
           output (new midi :time (now)
                        :keynum k
                        :duration (* rate 1.5)
                        :amplitude .5 )
           and wait rate))
```

The **`play-exp`** process passes the *a* stretching factor to the **`ran`** function. Since the distribution is unbounded the conditional clause insures that only valid MIDI key numbers will be output.

Interaction 18-5. Playing the exponential distribution for a=1.

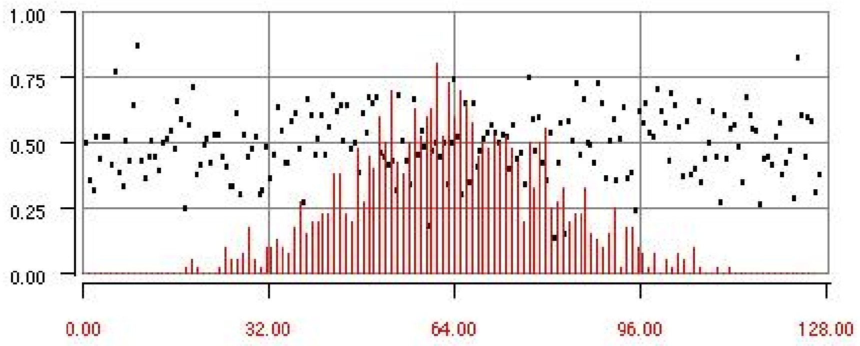
```
cm> (events (play-exp 50 .1 20 100 1) "exp.midi"
            :channel-tuning ct9)
"exp-1.midi"
cm>
```

- exp-1.midi

## Gaussian Distribution

The Gaussian, or normal, distribution returns values that fall under the normal curve. The spread, or *standard deviation* is 1.0 centered at 0, so 68.26% of the results are between -1 and 1, and 99.74% of the results are between -3 and 3.

Figure 18-6. Gaussian distribution values (black points) and histogram (red bars).



Example 18-3. A Gaussian distribution.

```
(define (play-gauss len rate key1 key2)
  (process repeat len
           for r = (ran :type :gaussian)
           for k = (rescale r 0.0 3 key1 key2)
           when (<= 0 k 127)
           output (new midi :time (now)
                        :keynum k
                        :duration (* rate 1.5)
                        :amplitude .5 )
           and wait rate))
```

Interaction 18-6. Playing Gaussian distribution.

```
cm> (events (play-gauss 50 .1 20 100) "gauss.midi"
            :channel-tuning ct9)
"gauss-1.midi"
cm>
```

- gauss-1.midi

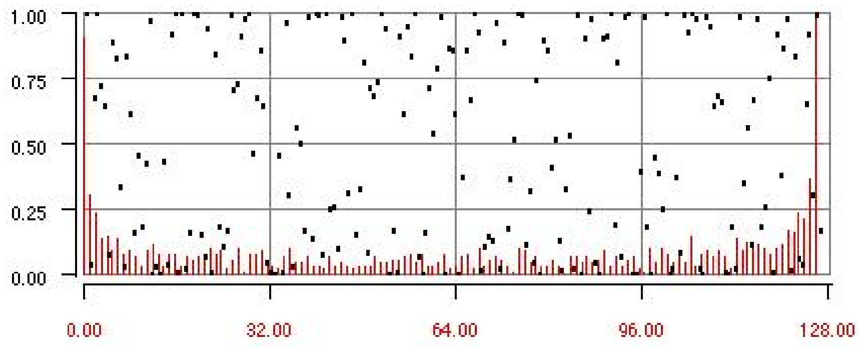
## Beta Distribution

The Beta distribution is one of the most musically interesting distributions to work with because it describes a whole family of distribution shapes. The beta distribution is controlled by two parameters *a* and *b*. When *a=b=1* then a uniform distribution results. When *a=b*, the distribution is symmetric around .5. When *a<1* and *b<1* the density of large and small numbers increases. When *a>1* and *B>1* density is similar to the Gaussian distribution.

Example 18-4. The beta distribution.

```
(define (play-beta len rate a b key1 key2)
  (process repeat len
           for r = (ran :type :beta :a a :b b)
           for k = (rescale r 0.0 1.0 key1 key2)
           output (new midi :time (now)
                        :keynum k
                        :duration (* rate 1.5)
                        :amplitude .5 )
           wait rate))
```

Figure 18-7. Beta distribution values (black points) and histogram (red bars).



Interaction 18-7. Playing the beta distribution for a=b=.4.

```
cm> (events (play-beta 50 20 90 .1 .4 .4 ) "beta.midi"
             :channel-tuning ct9)
"beta-1.midi"
cm>
```

- beta-1.midi

## Random Processes in Music Composition

Both continuous and discrete random processes are used in computer composition to control the evolution of parameterized data. But — as may be noticed in the preceding examples — simply mapping random distributions directly onto sound parameters does not translate into interesting musical results. The lack of correlation between events in a random process can cause our interest to diminish rapidly as we listen. There are really only two ways to deal with this issue. One method is to reserve the use of randomness for aspects in the composition that do not affect the structural integrity of the music. The other method is to control randomness in some manner such that the listener either perceives an underlying structure unfolding or at least has some expectation of regularity, or correlation in the pattern. The remaining sections of the chapter present different approaches to controlling random processes such that they generate more interesting musical results than the basic distributions on which they are based.

## Sampling Without Replacement

Generating discrete random values using an expression like **`(random 88)`** is called *sampling*. Sampling can be performed two different ways. *Sampling with replacement* means that once a value has been sampled it is still available to be sampled again. For example, if key number 60 is generated by **`(random 88)`** then 60 can be still be selected as a possible next value. *Sampling without replacement* describes a process in which generated values are not returned to the population once they have been selected. This is similar to how cards are dealt from a shuffled deck. By using sampling without replacement, a composer randomly reorders a set of specific values whose relationships are otherwise fixed. This reordering is perceived as a kind of freshness in the data. The function **`shuffle`** can be used to implement sampling without replacement. Shuffle randomly permutes a list of values, which can then be sampled one at a time to produce a random ordering without repetition. Since shuffle operates on list positions rather than values, the elements in the list do not have to be numeric.

Example 18-5. Sampling without replacement.

```
(define (sampling reps rate notes )
  (let ((len (length notes)))
    (process for i below reps
             for j = (mod i len)
             when (= j 0) ; need to reshuffle?
             set notes = (shuffle notes)
             output (new midi :time (now)
                         :keynum (list-ref notes j)
                         :duration (* rate 1.5)
                         :amplitude (interp j 0 .4 (- len 1) .8))
             wait rate)))
```

The **sampling** process produces a crescendo that starts over each time the values are shuffled so that each "period" in the population can be heard. Note that while shuffle avoids direct repetition within the period, there may still be a direct repetition between the last note of a period and the first note of the next.

Interaction 18-8. Calling shuffle.

```
cm> (events (sampling 80 .1 '(c d ef f g a bf c5))
            "shuffle.midi")
"shuffle-1.midi"
cm>
```

- shuffle-1.midi

Direct reselection can be an issue anytime discrete random selection is used. When the range of random values is sufficiently large direct repetition may be so infrequent that it does not have a noticeable effect on the results. But as the number of events in a discrete process diminishes the likelihood of direct reselection increases. At some point this may cause an unwanted regularity in musical results. Use the optional *omit* argument to the **between** function to avoid direct reselection. An *omit* value will not be returned by **between**. By specifying the last output value as the *omit* value for the next random number, direct reselection will not occur. The following process can be used experiment with the effects of allowing or inhibiting direct reselection.

Example 18-6. Avoiding direct repetition in discrete distributions.

```
(define (reps len lb ub skip?)
  (process repeat len
           for l = (if skip? k false)
           for k = (between lb ub l)
           output (new midi time (now)
                        :keynum k
                        :duration .2)
           wait .125))
```

If *skip?* is true then the variable *l* is set to *k*, the last value returned by **between**. The value **l** is passed as the omit value to **between** and so so direct repetition will occur. If *skip?* is false then *l* will be false no matter what the value of *k* and then direct selection is possible.

Interaction 18-9. Random selection with and without direct repetition.

```
cm> (events (reps 30 60 63 false) "reps.midi")
"reps-1.midi"
cm> (events (reps 30 60 63 true) "reps.midi")
"reps-2.midi"
cm>
```

- reps-1.midi
- reps-2.midi

## Variance

Continuous selection is commonly used as a means to produce variance in scaling and offsetting procedures. For example, by scaling a regular tempo or pulse value by a small random percentage the actual rate of each event will fluctuate and produce a more natural effect than a fixed rate. If the percentage variance is small the fluctuations will not affect the perception of an underlying structural pulse. The process definition in Example 18-7 can be used to experiment with random percentage variance applied to the pulse, amplitude and duration of events.

Example 18-7. The tapping process.

```
(define (tapping reps pct rate key amp)
  (let ((half (/ reps 2)))
    (process for i below reps
             for v = (if (< i half) 0 pct)
             output (new midi :time (now)
                          :duration (vary .1 v )
                          :keynum key
                          :amplitude (vary amp v :below))
             wait (vary rate v))))
```

The **`tapping`** process plays a single *key* number *reps* times at a specified *rate* and amplitude. The events in the first half of the process are played with fixed values and in the second half the values are allowed to vary by up to *pct* amount of their values. By default the function **`vary`** distributes variance equally on either side of the fixed value. To produce variance less than the value specify **`:below`** as the third argument to **`vary`**. Similarly, the keyword **`:above`** will place the variance above the value.

Interaction 18-10. Comparing 10% variance with 50% variance.

```
cm> (events (tapping 50 .1 .2 72 .6) "tap.midi")
"tap-1.midi"
cm> (events (tapping 50 .5 .2 72 .6) "tap.midi")
"tap-2.midi"
cm>
```

- tap-1.midi
- tap-2.midi

Even something as simple as random variation of a beat can be made musically interesting if it is controlled in some manner. In the next example the amount of variation applied to the beat is controlled by an envelope. When the envelope is at 0 then no variation occurs, when the envelope is at 1 then full variance is expressed. By using an envelope to control random variance the process can interpolate over the full continuum between completely random time increments to a measured pulse. The same envelope that controls the beat will also be used to specify a value for the *a* and *b* parameters to the beta distribution that generates amplitude values for the process. When the control envelope is at zero (no random variance) then the beta distribution will be *a=b=.4* and produce a large percentage of loud and soft values (Figure 18-7). When the envelope is at 1 then the beta distribution will be *a=b=1* and produce values in the uniform distribution. When the control envelope is at zero (no random variance) then the beta distribution will be at (beta *a=b=1*) and produce values in the uniform distribution. When the envelope is at 1 (full random variance) then *a=b=.4*) and a large percentage of loud and soft values are produced. The shape of the envelope defined in Example 18-8 causes the process to begin and end with random time values and express the strict pulse in the middle.

Example 18-8. The rain process.

```
(define (rain len rate env key1 key2 amp)
  (process for i below len
           for e = (interpl (/ i len) env)
           for v = (vary rate e :above)
           ;; rescale e to control shape of beta distribution
           ;; when e=0 z=.4, and when e=1 z=1.
           for z = (rescale e 0 1 1 .4)
           for b = (ran :type :beta :a z :b z)
           output (new midi :time (+ (now) v)
                       :duration v
                       :keynum (between key1 key2)
                       :amplitude (rescale b 0 1 .1 amp))
           wait rate))
(define rain-env '(0 1 .4 0 .6 0 1 1))
```

We listen to three simultaneous versions of the **`rain`** process, each with different length beats.

Interaction 18-11. Listening to rain.

```
cm> (events (list (rain 80 .5 rain-env 70 90 .8)
                  (rain 40 1 rain-env 40 60 .8)
                  (rain 20 2 rain-env 20 40 .9))
             "rain.midi")
"rain-1.midi"
cm>
```

- rain-1.midi

## Shape based composition

If a random variance is applied to envelope values we can generate random variations on the basic shape of the envelope:

Example 18-9. Adding a random variance to envelope values.

```
(define (shaper len rate env key1 key2)
  (process for i below len
           for e = (interpl (/ i len) env )
           for k = (rescale e 0 1 key1 key2)
           for r = (between -5 5)
           output (new midi :time (now)
                       :keynum (+ k r)
                       :duration rate)
           wait (* rate (pick .5 .25 1))))

(define shape-env
  '(0 0 .2 1 .4 .25 .6 .5 .8 0 1 .5))
```

The process in Example 18-9 interpolates values from an envelope and then scales and offsets them to lie within a range of key numbers specified to the process. A random amount is then added to the the key number to shift it upwards or downwards.

Interaction 18-12. Listening to two variations of the same envelope.

```
cm> (events (list (shaper 40 1 shape-env 70 90)
                  (shaper 40 1 shape-env 60 80)
                  (shaper 20 2 shape-env 60 48))
```

```
            "shape.midi")
"shape-1.midi"
cm> (events (list (shaper 40 1 shape-env 70 90)
                  (shaper 40 1 shape-env 60 80)
                  (shaper 20 2 shape-env 60 48))
            "shape.midi")
"shape-2.midi"
cm>
```

- shape-1.midi
- shape-2.midi

### Tendency Masks

A *tendency mask* is a technique related to the shape-based approach first described by the composer Gottfried Michael Koenig. A tendency mask is created by defining two envelopes that specify the lower and upper boundaries for a random selection. By describing each boundary as an envelope the range of the random variable can vary as a function of some dynamic process. Common Music's **`tendency`** function returns a random value given an *x* lookup value, and a lower and upper envelope that define the boundaries.

Example 18-10. Tendency masking.

```
(define low-mask '(0 .4 .75 .4 1 0))
(define top-mask '(0  1 .25 .6 1 .6))

(define (masker len rate lower upper key1 key2)
  (let ((last (- len 1)))
    (process for i to last
             for e = (tendency (/ i last) lower upper)
             for k = (rescale e 0 1 key1 key2)
             output (new midi :time (now)
                         :duration rate
                         :keynum k)
             wait rate)))
```

In this example two envelopes *lower* and *upper* control the range of random selection. The variable *e* is set to a random value between an interpolated lower and upper bounds in the two envelopes specified to **`tendency`** This random value, which lies between zero and one, is then rescaled to lie between the key numbers *key1* and *key2* passed as arguments to the process.

Interaction 18-13. Listening to masker.

```
cm> (events (masker 80 .1 low-mask top-mask 20 110)
            "masker.midi")
"masker-1.midi"
cm>
```

- masker-1.midi

## Weighted Random Selection

In *weighted random selection* individual probability weights are assigned to each event in a discrete random process. This affords more precise control over the overall mix of outcomes than would otherwise be possible using one of the distributions already discussed in this chapter. Common Music's **`odds`** function implements a simple binary form of weighted random selection that is identical to the **`choose?`** function from Chapter 5 except that values other than true and false can be returned by **odds**. More complex forms of weighted selection are presented in Chapter 19 and Chapter 20, but these discussions assume an understanding of probability tables, which we present here.

## Probability Tables

A probability table associates each outcome in a discrete process with a particular segment of the total probability. The size of each segment reflects the proportional likelihood of that value being generated from the random process. To generate the next value from a table, a uniformly distributed random value is generated and the table is searched to find the probability segment that contains the random lookup value. The datum associated with that segment is then returned from the table as the random value. Since the table operates on segments rather than values the weighted random process can generate non-numeric as well as numeric values. The implementation defined in Example 18-11 also allows us to specify probabilities in terms of relative *weights* that are associated with each event in the random process. The function that creates the probability table will convert these relative weights into absolute portions of the total probability interval 0 to 1. To see how this works, consider a weighted random process that generates four rhythmic symbols: S, E, Q and H where S should be three times as likely Q, E should be twice as likely as Q and H should 1/4 as likely as Q (Table 1).

Table 18-1. Example probability table for four rhythmic outcomes.

| | | | | |
|---|---|---|---|---|
| **index** | 0 | 1 | 2 | 3 |
| **outcome** | s | e | q | h |
| **weight** | 3 | 2 | 1 | 1/4 |
| **segment** | 0.0-0.48 | 0.048-0.8 | 0.8-0.96 | 0.96-1.0 |

Our implementation of probability tables (ptables) in Example 18-11 requires two functions. The first function **make-ptable** creates a ptable from a list of outcomes and weights. The **pran** function generates a random outcome from a ptable created by **make-ptable**.

Example 18-11. Implementing probability tables.

```
(define (make-ptable data)
  (let ((total (loop for d in data sum (second d)))
        (sum 0))
    ;; total holds sum of weights in data
    (loop for d in data
          for v = (first d)      ; outcome to return
          for w = (second d)     ; relative weight
          do (set! sum (+ sum w))
          ;; collect outcome and normalized probability
          collect (list v (/ sum total)))))

(define (pran table)
  ;; return outcome in table according
  ;; to its weighted probability
  (let ((x (random 1.0)))
    ;; x is uniform number < 1.
    (loop for d in table
          for p = (second d)
          when (< x p )   ; x in this segment.
          return (first d))))
```

A probability table is represented as a list of lists, the first element of each sub-list is an outcome and the second value is its weight relative to the other outcomes defined in the list. For example, the probability table described in

Table 1 would be specified as the list:

```
      ((s 3) (e 2) (q 1) (h 1/4))
```

The **`make-ptable`** function creates a probability table from *data*, a list of values and weights as shown in the example above. The table returned by **`make-ptable`** is identical to *data* except that the relative probability weights in the data description are converted to equivalent absolute positions along the probability interval 0-1. The variable *total* is set to the sums of the relative weights in *data*. The loop then iterates over each sub list *d* in *data* and sets the variable *v* to the outcome in *d* and sets *w* to its weight. The variable *s* is then set to the cumulative sum of each weight encountered so far. The loop collects lists containing the same value *v* but with *s* normalized to lie between 0-1. This is accomplished by calculating the current value of *s* to the *total* weight. The table returned by the function contains the same order of outcomes as *data* but with relative weighted converted to monotonically increasing probabilities that represents the upper bounds for segments along the probability range 0-1.

The function **`pran`** implements weighted random selection given a probability table. It generates a uniform random number *x* between 0 and 1. This value is used as a “lookup value” to find in the table. Since *x* is a uniform number, the larger segments in the table will contain proportionally more chances of containing values of *x* than the smaller segments. The loop iterates over each element in the table and compares *x* to the probability segment value *p* of each segment. Since segment values are stored in increasing order, the first occurrence of *x* less than *p* is the probability segment that contains the value to return. Given this implementation of probability tables we can now define a process to generate values using weighted random selection:

Example 18-12. Weighted random selection of notes and rhythms.

```
(define note-tabl
  (make-ptable '((c 3) (d 1) (ef 2) (f 1) (g 3)
                 (af 1) (bf 1) (c5 3) )))

(define rhy-tabl
  (make-ptable '((s 3) (e 2) (q 1)  (h 1/4))))

(define (play-pran reps notes rhys tempo)
  (process repeat reps for k = (keynum (pran notes))
           for r = (rhythm (pran rhys) tempo)
           output (new
                    midi :time (now) :keynum k
                    :duration (* r 1.5))
           wait r))
```

The probability table defined in *notes* “prefers” notes of a C-minor triad in the C-minor scale. The table stored in *rhys* contains the example shown in Table 1.

Interaction 18-14. Listening to Pran

```
cm> (events (play-pran 50 note-tabl rhy-tabl 100) "ptab.midi")
"ptab-1.midi"
cm>
```

- ptab-1.midi

## Random Walks

The term *random walk* describes a random process in which a value is continually incremented by a random amount. The motion produced is not unlike how a drunk staggers from point to point on each step: while the drunk's next position is not known exactly it will lie somewhere within the maximum span of one step. The *random walk* is characteristic of how molecules and very small particles move (Brownian motion) and is also referred to as *brown noise* for that reason.

Because the next value in a random walk must lie within a certain interval from the current value, the current value has a very strong influence on the next outcome. The influence of past values on successive values is called *correlation*. The correlation in a random walk is characterized by a $1/f^2$ spectrum, which means that small rates of change are much more present than larger ones. This is a very different behavior than the uniform distribution, or *white noise*, which exhibits large changes just as frequently as small ones because there is no correlation between successive values.

A random walk is easy to implement in a musical algorithm. A walking value *x* is initialized to some appropriate value and then incremented a random amount (positive or negative) on each iteration of the process. A moment's reflection will tell us that at some point an *unbounded* walk will likely exceed the limits of whatever sound parameter it is controlling. For this reason the random walks that control parameterized data are typically *bounded*, or constrained, in some manner so that they always lie within minimum and maximum boundary values. There are no hard and fast rules on how to adjust values that walk past the boundaries: common adjustments include resetting value to the boundary value, reflecting the value back into bounds, randomly repositioning the value somewhere within the range, or stopping the process altogether.

Common Music's **`drunk`** function implements bounded or unbounded random walks. Optional keyword arguments constrain values to lie within specified *:low* and *:high* boundaries. A value that exceeds either bounds is adjusted according to the *:mode* argument. Possible modes are:

```
:reflect :limit :reset :jump :stop
```

The default mode is **`:reflect`**, which moves a value back into bounds by its excess amount. Example 18-13 defines a process that uses **`drunk`** to generate a random melody.

Example 18-13. Random walks applied to keynum and amplitude.

```
(define (stagger len low high step)
  (let ((k (/ (- high low) 2))
        (a (between .1 .9)))
    (process repeat len
             for r = (pick 1/8 1/4)
             set k = (drunk k step :high .9 :mode :jump)
             output (new midi :time (now)
                         :duration (* r 1.5)
                         :keynum k :amplitude a)
             wait r)))
```

*K* is a key number controlled by a random walk between *key1 key2*. On each iteration of the process the key number can deviate by up to *step* steps from its current position. The variable *a* is the amplitude of each event, which is allowed to wander between .1 and .9. The **`:mode`** keyword argument controls the boundary rule of the walk. The value **`:jump`** causes a value that is out of bounds to be reset to a randomly chosen position within bounds.

Interaction 18-15. Listening to the stagger process.

```
cm> (events (stagger 50 40 80 3) "drunk.midi")
"drunk-1.midi"
cm> (events (stagger 50 20 90 7) "drunk.midi")
"drunk-2.midi"
cm>
```

- drunk-1.midi
- drunk-2.midi

## Example

The final example in this chapter was provided by Michael Klingbeil from some studies he made for an orchestra piece. The example generates a series of gestures, each of which consists of an upward "arc" of microtonal notes related by variation and exponentiation. This short example uses many of the functions presented in this chapter, the exponential scaling functions from Chapter 16 and the harmonic series from Chapter 15.

Example 18-14. Using randomness in scaling and tuning.

```
(define (arpeggiate-exprhy keynums time rate midpoint-frac
                           amplow amphi legato bass-legato
                           bass-cutoff last-legato)
  (let* ((segs (length keynums))
         (last (1- segs))
         (midpoint (floor (* segs midpoint-frac)))
         ;; deltas below midpoint follow one curve, above another.
         (deltas (append (explsegs midpoint (* midpoint-frac time)
                                   rate)
                         (explsegs (- segs midpoint)
                                   (* (- 1 midpoint-frac) time)
                                   (/ 1 rate)))))
    (process for i from 0
             for k in keynums
             for d in deltas
             for r = (if (< k bass-cutoff)
                       bass-legato
                       (if (= i last)
                         (* last-legato d)
                         (* legato d)))
             for a = (between 0.45 0.5)
             then (drunk a 0.1 :low amplow :high amphi )
             output (new midi :time (now)
                         :keynum k
                         :amplitude a
                         :duration r)
             wait d)))
```

The **`argeggiate-exprhy`** process creates both upward and downward "arcs" of MIDI events from a specified set of key numbers. The rhythms of each arc are determined by dividing a total time duration into two segments of exponentially related rhythms. The *midpoint-frac* is a proportion that determines the duration of each are relative to the total *time* specified to the process. This duration is then "chopped up" into exponentially related time values that sum to the duration of the arc. The amplitude *a* for the each event is determined by a random walk. The *amplow* and *amphi* parameters control the overall range of the random walk. *Bass-legato* sets the duration of notes with keynum less than *bass-cutoff.* This allows the bass notes to ring even when the synthesizer does voice stealing for the middle notes of the arpeggio. *Last-legato* extends the duration of the final note of an arpeggio for similar reasons.

The definition of **`argeggiate-exprhy`** produces a single upward and downward arpeggiation. We now define another process that schedules a series of overlapping arpeggiations whose key numbers are determined by distorting the harmonic series:

Example 18-15. Arpeggiated harmonic distortions.

```
(define (distort-harmonics fund distort)
  (loop for h from 1 below (floor (/ 25.0 distort))
        if (odds (* 0.9 distort))
        collect (keynum (* fund (expt h distort))
                        :hertz)))

(define (arpa-harmonic note dur gap)
  (process with fund = (hertz note)
           for distort from 0.7 below 1.05 by 0.05
           for notes = (distort-harmonics fund distort)
           sprout (arpeggiate-exprhy notes
                                     (* (vary dur 0.1) distort)
                                     (between 4.0 0.25)
                                     (between 0.3 0.6)
                                     0.3  ; amplow
                                     0.8  ; amphi
                                     (* dur distort 0.7) ; bass-legato
                                     2.0   ; legato
                                     59    ; bass cutoff
                                     1.0)  ; last-legato
           wait (vary gap 0.4)))
```

**`Arpa-harmonic`** spawns overlapping arpeggios with maximum mean duration of *dur* and a mean *gap* between arpeggio starts of *gap* seconds. Each arpeggio is composed of frequencies of a distorted harmonic series. The distortion factor varies from 0.7 (compressed spectrum) to 1.0 (harmonic spectrum) by increments of 0.05. The greater the compression, the lesser the odds of selecting an element of the spectrum. When the distortion reaches 1.0 the spectrum is harmonic and on average 90% of the elements will be selected.

Interaction 18-16. Arpeggiated harmonic distortions.

```
cm> (events (arpa-harmonic 'g1 7.0 5.0) "arp.midi"
            :channel-tuning ct9)
"arp-1.midi"
cm> (events (arpa-harmonic 'g1 6.0 5.0) "arp.midi"
            :channel-tuning ct9)
"arp-2.midi"
cm>
```

- arp-1.midi
- arp-2.midi

# Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file chance.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 19 Markov Processes

The previous chapter Randomness and Chance Composition introduced random number generation and presented simple control strategies that produce musical results more satisfying than simply mapping distributions directly onto sound parameters. Without additional controls probability distributions sound incoherent due to a lack of correlation between the generated samples. In the uniform distribution events have an equal likelihood of being selected and the choice of the previous value has no influence on the choice of the next. In non-uniform distributions, certain events have more probability than others but the choice of the next event is still independent of the previous choice. But in a process like the random walk there is a correlation between successive random events. Since a random walk is described by an increment, the position of the next event in the walk is strongly influenced by the current event.

A *Markov process*, or *Markov chain*, is a type of random process that is closely related to the random walk. In a Markov process past events represent a state, or context, for determining the probabilities of subsequent events. The number of past events that are used by the process is called the *order* of the Markov process. In a first order process the probabilities for choosing the next event depend only on the immediately preceding event. In a second order Markov process the probabilities for the next choice depend on the last two events. A Markov process can reflect any number of past choices, including the degenerate case of no past choices. A zero order Markov Process is equivalent to weighted random selection.

Because past choices influence subsequent choices, a Markov process can model, or mimic, patterns in data. The higher the order, the closer the process can come to matching a pattern it models. The pattern on which a Markov process is based can be invented by the composer or determined by a statistical analysis of data. This data need not be quantitative in nature. We will examine both methods for creating Markov processes later in this chapter.

## Conditional Probability and Transition Tables

A Markov process exhibits *conditional probability*, where the probability of the next outcome in depends on one or more past outcomes. This means that the probability distribution of the random process will change dynamically as a function of the outcomes that the process has already produced. Conditional probability is implemented in a Markov process by linking past outcomes with specific probability distributions for choosing the next event. For two independent events $x$ and $y$, the conditional probability P($x$|$y$) is equal to the product of the probability of $x$ and $y$ alone. For example, the probability of "heads" appearing in a coin toss is P($h$)=.5. The conditional probability of consecutive heads P($h$|$h$) is equal to the products .5*.5=.25. The probability of tossing three heads is .5*.5*.5=.125 and so on. Higher order processes can be described by multiplying the probabilities stored in a first order table or by explicitly storing past outcomes in a table and then matching these against a *history* of outcomes to determine the probability of future outcomes. The table that establishes this linkage is called a *transition table*. A transition table is really a table of probability tables. Each transition represents the probability distribution for a specific set of past outcomes. In a first order Markov process there is only one past outcome in a transition. In a second order process each transition would refer to a pair of past outcomes.

Transition tables are similar to the probability tables discussed in Chapter 18 but are representationally more powerful structures. We can show the relationship between probability tables and transition tables by transforming the probability table shown in Table 19-1 into a transition table and then implementing behavior that is not possible with a probability table alone. Table 19-1 is a probability table for a fair coin toss. There are only two possible outcomes, **h** (heads) and **t** (tails), each of which has an equal probability of being selected:

Table 19-1. Probability table for a fair coin toss. Heads is labeled **h** and tails **t**.

| h | t |
|---|---|
| 0.50 | 0.50 |

Table 19-1 can be turned into a transition table by defining specific *transitions* for the two possible outcomes. A transition links past outcomes with future probabilities. In the case of our table there are only two past outcomes possible: `h` and `t`. Each row in a transition table implements one transition rule. The columns in the table provide all the possible outcomes for each transition. Each cell in a transition row represents the probability of the transition producing a specific future outcome. Table 19-2 defines a transition table for a fair coin toss. A Markov *process* represents an implicit flow of outcomes through the table: random outcomes are generated from the top of the table (labeled next in Table 19-2) and then “fed back” into the table as the transition rows (marked past):

Table 19-2. Transition table for a fair coin toss.

| | | ***next*** | |
|---|---|---|---|
| | | **h** | **t** |
| ***past*** | **h** | 0.50 | 0.50 |
| | **t** | 0.50 | 0.50 |

The random outcomes produced by Table 19-2 have the same probability distribution as in Table 19-1. The following sequence depicts 20 representative outcomes:

t h t t h h h t t t h h t h h h t h t t

The next two examples demonstrate two different distributions that are impossible with the probability table in Table 1. Each of these new distributions is created by specifying different probability value in the cells without altering the structure of the table. By simply changing the weights in the table's cells different coin tossing behavior can be implemented. The next table creates a biased coin toss which always favors heads, particularly if the last value was tails:

Table 19-3. Transition table for a biased coin toss.

| | | ***next*** | |
|---|---|---|---|
| | | **h** | **t** |
| ***past*** | **h** | 0.75 | 0.25 |
| | **t** | 1.00 | 0.00 |

Representative outcomes from Table 19-3:

t h t h h h h h t h h t h h h h h t h h

The weights in Table 19-3 cause tails to always produces heads, and heads to usually produces heads. In Table 19-4 we implement a type of coin toss in which the current outcome (heads or tails) tends to persist.

Table 19-4. Transition table for a persistent coin toss.

| | | ***next*** | |
|---|---|---|---|
| | | **h** | **t** |
| ***past*** | **h** | 0.60 | 0.40 |
| | **t** | 0.20 | 0.80 |

Representative outcomes from Table 19-4:

t t h t h t h h h t t t t t t t t h t t

## Representing Transition Tables

Transition tables can be represented by adapting our design for probability tables from Chapter 18. Recall that the constructor function **`make-ptable`** created a probability table from a list of data that associated random events with their relative weights in the table. We start by observing what these probability lists would look like for the transitions in Table 4:

Table 19-5. Probability lists for Table 4.

| | **h** | **t** |
|---|---|---|
| **transition 1** | `((h .6)` | `(t .4))` |
| **transition 2** | `((h .2)` | `(t .8))` |

To turn these probability lists into transition lists only requires that we explicitly associate a past outcome with each set of probability pairs. We can do this by adding the past outcome to the front of each probability list:

Table 19-6. Transition lists for Table 4.

| | **past** | **h** | **t** |
|---|---|---|---|
| **transition 1** | `(h` | `(h .6)` | `(t .4))` |
| **transition 2** | `(t` | `(h .2)` | `(t .8))` |

Each transition list is now a heterogeneous structure linking a past outcome with a probability distribution for generating the next outcome. The first element in each transition is the past outcome and the rest of the elements in the list are the probability pairs. Note that to implement a second order table each past value would be a list of past values. For example, a second order representation for Table 4 might look like:

Table 19-7. Second order transition lists.

| | **past** | **h** | **t** |
|---|---|---|---|
| **transition 1** | `((h h)` | `(h .9)` | `(t .1))` |
| **transition 2** | `((h t)` | `(h .9)` | `(t .1))` |
| **transition 3** | `((t t)` | `(h .9)` | `(t .1))` |
| **transition 4** | `((t h)` | `(h .9)` | `(t .1))` |

Notice that as the order of the Markov process increases so does the length of the transition table since each possible combination of past outcomes must be explicitly present in the table.

## Implementing Transition Tables

In our implementation of transition tables, each **`ttable`** will consist of a list of transitions and each transition will be a list that associates one or more past outcomes with a ptable (probability table) for generating the next outcome (Table 6.) We will define three functions to implement ttables and Markov selection:

1. The constructor **`make-ttable`** creates a ttable from an external specification of its data.
2. The accessor **`find-ptable`** finds the ptable for a specified past outcome in a ttable.
3. The Markov selection function **`mran`** returns the next random outcome from a ttable and past outcome.

Our implementation assumes that we have already defined the required ptable definitions presented in Example 18-11 of the previous chapter. We start by defining the constructor function for transition tables:

Example 19-1. Defining a transition table constructor.

```
(define (make-ttable data)
  ;; create a transition table from data.
  (loop for trans in data
        for past = (first trans)
        for probs = (rest trans)
        ;; each row transition is a past value and a ptable
        collect (list past (make-ptable probs))))
```

The function **`make-ttable`** creates a ttable from a list of *data*. Each element in *data* is a transition list *trans*. The first element in each *trans* is a past outcome and the rest of the data are the probability pairs *probs* that make up its distribution. The **`make-ttable`** returns a ttable list as its value. Each list in a ttable is a transition list that holds a past outcome and its associated probability table created by *make-ptable*.

Interaction 19-1. Creating a transition table from Table 4.

```
cm> (define coin
      (make-ttable '((h (h .6) (t .4))
                     (t (h .2) (t .8)))))

cm> (pprint coin)
((h ((h 0.6) (t 1.0)))
 (t ((h 0.2) (t 1.0))))
cm>
```

The next step is to define the function **`find-ptable`** which returns a **`ptable`** for a specified past outcome. This function will accept two arguments, the transition table *ttable* and the past outcome *past*, and will return the ptable associated with *past* in the transition table. The function will find the ptable by iterating over each transition and comparing each transition's past with *past*. If they are the equal then the ptable for that transition is returned. But what does "equal" actually mean? The values we must compare might be numbers, symbols (as is the case in our example table) or lists of data. How to we test the equivalence of two symbols or lists? The Lisp function **`equal?`** is a predicate for testing general equivalence between two Lisp objects:

Interaction 19-2. The equal? general equivalence predicate.

```
cm> (equal? 1 1)
#t
cm> (equal? 1 1.0)
#f
cm> (equal? 'a 'a)
#f
cm> (equal? '(t t) '(t t))
#t
cm> (equal? '(h t) '(t h))
#f
cm>
```

As a general rule of thumb, two objects are considered **`equal?`** if they look the same, otherwise they are not equal. Given the **`equal?`** general equivalence predicate we can define the probability table lookup function:

Example 19-2. A lookup function to find the ptable for a specified past outcome.

```
(define (find-ptable ttable past)
  ;; find the transition matching past and return its ptable
  (loop for trans in ttable
        for this = (first trans)
        if (equal? this past)
        return (second trans)))
```

The function **`find-ptable`** returns the probability table in *ttable* whose past outcome is equal to *past*. A loop iterates over each transition *trans* in the table and compares *past* with the past outcome in the first position of each transition. If these outcomes are the same, or **`equal?`**, then the loop immediately returns the probability table that is stored as the second element of *trans*. Note that the case of not finding a row for a past outcome should not happen. If it does (through mistaken input by the caller) the function will return false.

Interaction 19-3. Looking up ptables for past outcomes.

```
cm> (find-ptable coin 't)
((h 0.2) (t 1.0))
cm> (find-ptable coin 'h)
((h 0.6) (t 1.0))
cm> (find-ptable coin 'xyz)
#f
cm>
```

The final step in the implementation is to define the function **`mran`** that implements Markov selection. The function is passed two inputs, a *ttable* and a *past* outcome and returns the next outcome:

Example 19-3. Defining the Markov selection function.

```
(define (mran ttable past)
  ;; find row matching past and return its ptable
  (let ((ptab (find-ptable ttable past)))
    (if ptab
      (pran ptab)
      #f)))
```

We can now use Markov selection to return values from our "coin":

Interaction 19-4. Two methods for Markov selection.

```
cm> (loop with past = 'h
          repeat 20
          for y = (mran coin past)
          collect y
          do (set! past y))
(h h t t t h h h h t t t t t h h t t t t)
cm> (loop repeat 20
          for x = 'h then (mran coin x)
          collect x)
(h h t t t t t h h t t h h t t t h h h t)
cm>
```

Example 19-3 demonstrates two different strategies for controlling a Markov selection process. The first loop initializes *past* to one of the possible outcomes in the table. Each iteration of the loop then generates a value from the transition table and updates *past* with that value so that the next time through the loop that value is passed to **`mran`** as the past outcome. In the second loop the collection starts with a particular value and then updates that value on each subsequent iteration. In addition to being simpler than the first loop, the second method also allows the composer to specify a particular initial value for the beginning of the process. Other strategies for controlling the starting and ending points of Markov processes will be presented in the next few examples.

## Musical Examples

We will now use our implementation of transition tables to use first order Markov in a several musical contexts. For examples of higher order Markov processes please see the discussion of the Markov pattern in Chapter 20.

### Markov Rhythmic Patterns

Imagine generating a melody for a soloist in which rhythmic values are determined by random selection. Even if only very simple rhythms are used a moments reflection should tell us that the patterns produced by the process will not reflect any underlying pattern of beat. Assume that the process is restricted to quarters, dotted-eights, eighths, and sixteenths (which we notate q, e., e, and s respectively). Since the random process can place sixteenths and dotted eighths anywhere within a beat, the sequence of rhythmic values that result will only occasionally line up with the start of a metric pulse:

- uni-1.midi Uniform rhythmic distribution with tempo curve and random walk melody.

In this first Markov example we generate random rhythms that clearly express an underlying beat and tempo curve. A random walk is used to generate the melody.

Example 19-4. Comparison of uniform and Markov rhythmic patterns.

```
(define rhy-table
  '((q  (q .5) (e  2) (e. .75) )
    (e  (e 3) (q 1) )
    (e. (s 1))
    (s  (e 2) (q 1))))

(define tcurve '(0 1 .7 .75 1 1))  ; tempo envelope

(define (markov-rhythms len)
  ;; rhythms generated from first order Markov
  (let ((tbl (make-ttable rhy-table)))
    (process for i below len
             for k = 60 then (drunk k 6 :low 40 :high 80
                                    :mode :jump)
             for r = 'q then (mran tbl r)
             for d = (* (rhythm r 120)
                        (interpl (/ i len) tcurve))
             output (new midi :time (now) :keynum k
                        :duration d)
             wait d)))
```

The Markov table *rhy-table* assigns most weight to eighth notes and the least to dotted eighths. If a dotted-eighth is played then the next value must be a sixteenth, which forces the pattern to return back to the beginning of the beat again.

Interaction 19-5. Listening to Markov rhythm patterns.

```
cm> (events (markov-rhythms 80) "rhy.midi")
"rhy-1.midi"
cm>
```

- rhy-1.midi

The next version of the rhythmic process demonstrates an alternate method for controlling the starting and stopping conditions of a Markov process. The table in Example 19-5 encodes two extra transitions that represent unique states in the process. An initial state **start** for the table is represented by the first transition in the table. The output from this transition can be either a dotted-eighth or a quarter. A stopping state for the table is represented by the **stop** outcome that is the only outcome of the new whole note (w) transition. If a whole note is selected then the table will return the unique outcome **stop** as its next value.

Example 19-5. Markov rhythm with start and stop transitions.

```
(define rhy-table2
  '((start (e. 2) (q 1))
    (q  (q .5) (e  2) (e. .75) (w .2))
    (e  (e 3) (q 1) )
    (e. (s 1))
    (s  (e 2) (q 1))
    (w  (stop 1))))

(define (markov-rhythms2 )
  ;; rhythms generated from first order Markov with
  ;; explicit start and stop transitions.
  (let ((tabl (make-ttable rhy-table2))
        (past 'start))
    (process for k = 60 then (drunk k 6 :low 40 :high 80
                                    :mode :jump)
             for r = (mran tabl past)
             until (equal? r 'stop)
             for d = (rhythm r 120)
             output (new midi :time (now) :keynum k
                         :duration d)
             wait d
             set past = r)))
```

The process **markov-rhythms2** checks to see if *r* is equal to **stop**, in which case the process immediately terminates. One advantage with this method is that an ending value (in this case a whole note) can be expressed. The disadvantage with this method is that there is no way to determine ahead of time how long the process will actually run!

Interaction 19-6. Listening to Markov rhythm patterns.

```
cm> (events (markov-rhythms2 ) "rhy.midi")
"rhy-2.midi"
cm>
```

- rhy-2.midi

## Markov melody

In our next example we use a Markov process to generate a pseudo Gregorian Chant. The intention is not to generate a stylistically correct chant, but rather to see how melodic motion can be constrained using a Markov chain and how the processes can mimic composition styles. For a much larger example of style generation see the Markov process example in chapter Chapter 20. that implements the Stephen Foster style example described in Computer Music

In the crudest of terms, Gregorian Chant is characterized by mostly step wise motion within a range of modal degrees. From any given tone there is more likelihood that a chant will move a step up or down than then to skip a third or fourth away. The larger the skip, the more unlikely the chant is to make it. In an authentic mode, the final of the mode acts as a kind of “reflecting boundary” at the octave that redirects the melody in the opposite direction. Certain tones, such as the final and tenor, have more influence over the melody than others. In the Dorian mode the tenor A is occasionally decorated by the B-flat directly above it. This B-flat almost always returns back to the tenor tone. We can mimic these stylistic tendencies using a first order Markov process:

Example 19-6. Dorian Gregorian Chant

```
(define dchant
  '((d4  (d4 .1) (e4 .35) (f4 .25) (g4 .1) (a4 .15))
    (e4  (d4 .35) (f4 .35) (e4 .1) (g4 .1) (a4 .1))
    (f4  (d4 .2) (e4 .2) (f4 .1) (g4 .2) (a4 .12))
    (g4  (d4 .2) (e4 .1) (f4 .3) (g4 .1) (a4 .3) (bf4 .2))
    (a4  (d4 .1) (e4 .2) (f4 .25) (g4 .3) (a4 .1) (bf4 .3))
    (bf4 (a4 1))))

(define (monks1 len chant rhy)
  (let ((tbl (make-ttable chant)))
    (process repeat len
             for k = 'd4 then (mran tbl k)
             output (new midi :time (now)
                          :keynum k duration rhy)
             wait rhy)))
```

In Example 19-6 the Gregorian Chant is represented by a transition table that prefers step wise motion over leaps and repeated tones. Certain leaps, for example to the B-flat , are not allowed at all. B-flat can only be approached and left from the tenor note. The final **d4** is a reflecting boundary that most often move the melody up by step but occasionally to the mediant or tenor tones as well.

Interaction 19-7. Listening to Markov monks.

```
cm> (events (monks1 30 dchant .8) "monks.midi")
"monks-1.midi"
cm>
```

- monks-1.midi

In this next example we look at strategies for shaping a Markov process so that the results are more musically satisfying. We know, for example, that in Gregorian Chant the tenor and final are more likely to be agogically stressed then other tones. Moreover, the chant should begin and end on the final of the mode.

Example 19-7. Markov with conditional checks.

```
(define (chant-dur tone dur)
  ;; adjust dur if tone is D, F or A.
  (let ((doub (* dur 2)))
    (if (scale= tone 'd4)
      (odds .7 doub dur)
      (if (scale= tone 'a4)
        (odds .5 doub dur)
        (if (scale= tone 'f4)
          (odds .25 doub dur)
          dur)))))
```

```
(define (monks2 end chant rhy oct)
  (let ((tabl (make-ttable chant)))
    (process for k = 'd4 then (mran tabl k)
             for dur = (chant-dur k rhy)
             output (new midi :time (now)
                         :keynum (transpose k oct)
                         :amplitude .8
                         :duration dur)
             wait dur
             until (and (> (now) end)
                        (scale= k 'd4)))))
```

The **monks2** process generates a note *k* from a first order Markov chant and then calls **chant-dur** to determine the rhythmic value for that tone. **chant-dur** tests *k* to see if it is the final (d), tenor (a) or mediant (f) of the mode. If *k* is the final then there is a 70% chance that its duration will be doubled. Otherwise, if the tone is the tenor then there is a 50% chance the tone will be doubled. Otherwise, if the tone is the mediant **f4** then there is a 25% chance the duration will be doubled. The **until** clause at the end of the **monks2** process stops if the current note is the mode's final and enough notes have been sung. The *oct* parameter to the process allows us to shift each monk up and down in the register without having to alter the process definition. We can now listen to monks in octaves or in other interval relations.

Interaction 19-8. Listening to Monks2.

```
cm> (events (list (monks2 24 dchant .8 12)
                  (monks2 24 dchant .8 0)
                  (monks2 24 dchant .8 -12))
            "monks.midi")
"monks-2.midi"
cm> (events (list (monks2 24 dchant .8 -6)
                  (monks2 24 dchant .8 0)
                  (monks2 24 dchant .8 11))
            "monks.midi")
"monks-3.midi"
cm>
```

- monks-2.midi
- monks-3.midi

## Markov Harmonic Rules

In this next example we use a Markov process to generate the interval relations for a series of chords. The same rules will be used to generate the underlying melody onto which the chords are transposed.

Example 19-8. Markov interval patterns.

```
(define int-mix
  '((1 (3 .4) (4 .4) (6 .1))
    (2 (2 .2 ) (3 .4) (4 .4) (6 .1))
    (3 (1 .2 )(2 .6 )  (4 .4) )
    (4 (2 .2 ) (3 .4) (4 .4) )
    (6 (2 .4) (3 .2) (4 .2))))

(define (markov-chorder len int-mix note size ud rhy dur )
  (let ((tabl (make-ttable int-mix))
        (int 1)
        (key (keynum note))
        (chord #f))
    (process repeat len
             set int = (mran tabl int)
             set key = (fit (transpose key (odds ud int (- int)))
                            50 90)
             set chord = (loop with n = key
                               repeat size
                               collect n
                               do
                               (set! int (mran tabl int))
                               (set! n (transpose n (- int))))
             each c in chord
             output (new midi :time (now)
                          :keynum c duration dur)
             wait rhy)))
```

The **`markov-chorder`** process uses the interval content defined in *int-mix* to generate *len* number of chords. Each chord has *size* number of notes. The starting point for the series of chords is specified by *note* and *ud* is a ratio controlling the odds of the melody wandering upward or downward where a *ud* value greater than .5 prefers ascending melodic motion. The *int* variable will be used to store the current value from the Markov table and is initialized to 1. This value is then updated to a new interval from the Markov table at the beginning of each iteration of the process. The *key* variable serves as the key number on which each chord is based and its value is calculated by transposing the current value up or down *int* intervals according to the *ud* probability factor. The **`fit`** function insures that this value lies between key numbers 50 and 90. The next clause sets the value *chord* to a list of key numbers based on the current value of *key*. These key numbers are computed from the current key and the Markov interval table. The chord loop initializes the variable *n* to the current value of *key* and then iterates *size* number of times. On each iteration the value of n is first collected and then transposed downward by each interval read from the Markov table. (The transposition is forced downward simply to allow us to hear the melodic tone on which the chord is based more clearly.) The **`each`** process clause then applies an **`output`** action clause to each element in the chord.

Interaction 19-9. Listening to Markov generated harmony.

```
cm> (events (markov-chorder 25 int-mix 'g4 6 .6 1.2 1.2 )
            "chord.midi")
"chord-1.midi"
cm>
```

- chord-1.midi

## Markov Texture

In this last example we use a Markov process to generate a melodic texture that reminiscent of both the persistent coin toss model in Table 4 and the Gregorian Chant process discussed earlier in this section. In this example “heads” is represented by black keys and “tails” by white keys. Each black or white key in or ttable is weighted to prefer moving to nearer tones of the same color. The basic weighting strategy for moving from key to key is shown in Table 19-8

Table 19-8. Transition categories and weight spread.

| | |
|---|---|
| **Repeat current key:** | 0.5 |
| **Near-to-far spread same color** | 2.0-0.5 |
| **Near-to-far spread other color** | 0.2-0.1 |

Example 19-9. Persistent black and white keys.

```
(define bw-intervals
  ;; first line of each transition hold weights
  ;; for keys of same color. second lines contains
  ;; weights for moving to other color
  '((0  (0 .5) (2 2) (4 1.5) (5 1) (7 .5) (9 .5) (11 .5)
        (1 .2) (3 .1) (6 .1)  (8 .1) (10 .1))
    (1  (1 .5) (3 2) (6 1.5) (8 1) (10 1)
        (0 .2) (2 .2) (4 .1)  (5 .1) (7 .1) (9 .1) (11 .1))
    (2  (0 2)  (2 .5) (4 2)   (5 1.5) (7 1) (9 .5)
        (1 .2) (3 .2) (6 .1) (8 .1) (10 .1) (11 .1) )
    (3  (1 2)  (3 .5) (6 1.5) (8 1) (10 .5)
        (0 .1) (2 .2) (4 .2) (5 .1) (7 .1) (9 .1) (11 .1))
    (4  (0 1.5)(2 2)  (4 .5) (5 2) (7 1.5) (9 1) (11 .5)
        (1 .1) (3 .2) (6 .2) (8 .1) (10 .1))
    (5  (0 1)  (2 1.5)(4 2) (5 .5) (7 2) (9 1.5) (11 1)
        (1 .1) (3 .2) (6 .2) (8 .1) (10 .1))
    (6  (1 1.5)(3 2)  (6 .5) (8 2) (10 1.5)
        (0 .1) (2 .1) (4 .1) (5 .2) (7 .2) (9 .1) (11 .1))
    (7  (0 .5) (2 1)  (4 1.5) (5 2) (7 .5) (9 2) (11 1.5)
        (1 .1) (3 .1) (6 .2) (8 .2) (10 .1))
    (8  (1 1)  (3 1.5)(6 2) (8 .5) (10 2)
        (0 .1) (2 .1) (4 .1) (5 .1) (7 .2) (9 .2) (11 .1))
    (9  (0 .5) (2 .4) (4 1) (5 1.5) (7 2) (9 .5) (11 2)
        (1 .1) (3 .1) (6 .1) (8 .2) (10 .2))
    (10 (1 .5) (3 1) (6 1.5) (8 2) (10 .5)
        (0 .1) (2 .1) (4 .1) (5 .1) (7 .1) (9 .2) (11 .2))
    (11 (0 .5) (2 .5) (4 .5) (5 1) (7 1.5) (9 2) (11 0.5)
        (1 0.1) (3 .1) (6 .1) (8 .1) (10 .2))))

(define bw-octaves
  '((c3 (c3 2)   (c4 1)  (c5 .5) (c6 .25))
    (c4 (c3 1)   (c4 2)  (c5 1)  (c6 .5))
    (c5 (c3 .5)  (c4 1)  (c5 2)  (c6 1))
    (c6 (c3 .25) (c4 .5) (c5 1)  (c6 2))))

(define (bw len oct rate)
  (let ((ints (make-ttable bw-intervals))
        (octs (make-ttable bw-octaves))
        (reps 0)
        (int 0))
    (process repeat len
             if (= reps 0)
             set reps = (pick 4 8 12 16)
             and set oct = (mran octs oct)
             set int = (mran ints int)
             output (new midi :time (now)
                         :keynum (transpose oct int)
                         :duration (* rate 1.5))
             wait rate
             set reps = (- reps 1))))
```

The **`bw`** process uses two Markov tables to project an interval pattern of black and white key motion onto a pattern of octave transpositions. The Markov table *octs* that controls the octave register “prefers” motion to closer octaves than to octaves further away, which causes the octave register to generally shift in a smooth manner. The *reps* variable holds the number of notes remaining to play in the current octave. If *reps* is zero then a new octave is selected from *octs* and *reps* is reset to a random length of 4, 8, 12, or 16 events. The key number for each event is determined by transposing the current octave offset *oct* by an interval *int* generated from the Markov interval table *ints*. The last statement in the process decrements the value of *reps* by 1.

Interaction 19-10. Listening to Black and White.

```
cm> (events (bw 120 'c4 .125) "bw.midi")
"bw-1.midi"
cm> (events (list (bw 120 'c4 .125)
                  (bw 120 'c3 .125))
            "bw.midi"         '(0 4))
"bw-2.midi"
cm> (events (list (bw 120 'c4 .125)
                  (bw (* 120 21/13)
                      'c4
                      (* .125 13/21)))
            "bw.midi")
"bw-3.midi"
cm>
```

- bw-1.midi
- bw-2.midi
- bw-3.midi

The third call in Example 19-9 creates two rhythmic strata related by the Golden Mean ratio 21/13.

## Markov Analysis

The weights for all of the preceding Markov processes were determined only by the preference of the author. But Markov processes can also model the behavior of real world systems by analyzing the data and extracting a set of outcomes and weights that can then be used to regenerate chains with the same characteristics. To analyze the data a "window" is positioned over successive element in the sequence. The width of this window is determined by the order of the analysis. An analysis for a first order table uses a window width of 2: the first element represents a past outcome and the second is its successor. In a second order analysis the window is 3 elements wide, the first two elements are the past outcomes and the third element is that past's successor. The analysis proceeds iteratively, by moving the window sequentially over the elements and counting the number of times each unique outcome happens for each unique past. We can perform an example analysis on the following short sequence of data:

a a b c b a

A first order analysis proceeds uses a 2 element window, the first element in the pair represents a past value and the second element is its outcome. Windowing over the example sequence produces the following analysis pairs:

aa, ab, bc, cb, ba, aa

The very last window in the analysis (aa) is produced by wrapping the final window around to the front of the sequence again so that the last element has a successor. An analysis that wraps around creates a “seamless” chain without any terminating condition. An alternate method of analysis could encodes a unique terminal value, call it z, as the consequent of the last a. When z appears from the chain the caller will knows that the Markov process cannot generate any more events.

The next step in the analysis is to count each unique outcome for each unique past, which gives us the raw tally shown in Table 19-9. Past events are represented as row labels and successor events are labeled by columns. Table 19-10 shows the raw counts normalized to probability proportions.

Table 19-9. Raw count tally.

| | **a** | **b** | **c** |
|---|---|---|---|
| **a** | 2 | 1 | — |
| **b** | 1 | — | 1 |
| **c** | — | 1 | — |

Table 19-10. Normalized to proportions.

| | **a** | **b** | **c** |
|---|---|---|---|
| **a** | 0.667 | 0.333 | — |
| **b** | 0.500 | — | 0.500 |
| **c** | — | 1.000 | — |

It should be obvious even from this short example that doing a Markov analysis "by hand" is both tedious and prone to error. Luckily it is also a iterative task and something that computers can easily do!

## Analysis By Computer

Common Music provides the function **`markov-analyze`** to generate *nth* order Markov analysis of a sequence of data.

[Function]

`(`**`markov-analyze`** *seq* `[:order` *i*`] [:print?` *b*`] [:pattern?` *b*`] [:key` *p*`])`

The type of results returned by the function vary according to its **`:print?`** and **`:pattern?`** keyword arguments. One or more of the follow results are possible:

1. A printed table of results from the statistical analysis.
2. A pattern definition that describes the pattern.
3. A pattern object that implements the pattern.

Only the first and third results will be demonstrated in this chapter, see the discussion of the Markov pattern in Chapter 20 for information about the second option.

The following examples demonstrate different orders of analysis using the melody Happy Birthday. Happy Birthday is used because its structure is so well known by the listener that the effects of the musical Markov patterns will be clear in the examples. The effects of Markov on Happy Birthday was demonstrated to me by the composer Paul Berg, who works at the Institute of Sonology. The analyses shown below include tables for only the first two orders. Higher orders of analysis generate tables too large to fit on a book page but these can still be generated interactively in Lisp and then studied. Happy Birthday is fully expressed by a fifth-order Markov process. We first define the data sequence to analyze and a process to play the Markov renditions:

Example 19-10. Happy Birthday, data and player.

```
(define bday
  (note '(c4 c d c f e c c d c g f
          c c c5 a4 f e d bf bf a f g f)))

(define (play-bday order reps rate)
  ;; process to play happy birthday in various orders.
  (let ((pat (markov-analyze bday :order order
                             :print? false)))
    (process repeat reps
             output (new midi :time (now)
                         :keynum (next pat)
                         :duration (* rate 1.5))
          wait rate)))
```

### Happy Birthday, Zero-Order Markov (weighted random selection)

```
(markov-analyze bday :order 0 :print? :table :pattern? false)
```

Table 19-11. Zeroth order transition weights.

| c4 | d4 | e4 | f4 | g4 | a4 | bf4 | c5 |
|---|---|---|---|---|---|---|---|
| 0.32 | 0.12 | 0.08 | 0.20 | 0.08 | 0.08 | 0.08 | 0.04 |

Interaction 19-11. Listening to zero order Happy Birthday.

```
cm> (events (play-bday 0 40 .15) "bday.midi")
"bday-1.midi"
cm>
```

- bday-1.midi

### Happy Birthday, First Order Markov

```
(markov-analyze bday :order 1 :print? :table :pattern? false)
```

Table 19-12. First order transition weights.

| | c4 | d4 | e4 | f4 | g4 | a4 | bf4 | c5 |
|---|---|---|---|---|---|---|---|---|
| **c4** | 0.38 | 0.25 | — | 0.13 | 0.13 | — | — | 0.13 |
| **d4** | 0.67 | — | — | — | — | — | 0.33 | — |
| **e4** | 0.50 | 0.50 | — | — | — | — | — | — |
| **f4** | 0.40 | — | 0.40 | — | 0.20 | — | — | — |
| **g4** | — | — | — | 1.00 | — | — | — | — |
| **a4** | — | — | — | 1.00 | — | — | — | — |
| **bf4** | — | — | — | — | — | 0.50 | 0.50 | — |
| **c5** | — | — | — | — | — | 1.00 | — | — |

Interaction 19-12. Listening to first order Happy Birthday.

```
cm> (events (play-bday 1 40 .15) "bday.midi")
"bday-2.midi"
cm>
```

- bday-2.midi

### Happy Birthday, Second Order Markov

```
(markov-analyze bday :order 2 :print? :table :pattern? false)
```

Table 19-13. Second order transition weights.

| | c4 | d4 | e4 | f4 | g4 | a4 | bf4 | c5 |
|---|---|---|---|---|---|---|---|---|

| | | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| **c4 c4** | — | 0.67 | — | — | — | — | — | 0.33 |
| **c4 d4** | 1.00 | — | — | — | — | — | — | — |
| **c4 f4** | — | — | 1.00 | — | — | — | — | — |
| **c4 g4** | — | — | — | 1.00 | — | — | — | — |
| **c4 c5** | — | — | — | — | — | 1.00 | — | — |
| **d4 c4** | — | — | — | 0.50 | 0.50 | — | — | — |
| **d4 bf4** | — | — | — | — | — | — | 1.00 | — |
| **e4 c4** | 1.00 | — | — | — | — | — | — | — |
| **e4 d4** | — | — | — | — | — | — | 1.00 | — |
| **f4 c4** | 1.00 | — | — | — | — | — | — | — |
| **f4 e4** | 0.50 | 0.50 | — | — | — | — | — | — |
| **f4 g4** | — | — | — | 1.00 | — | — | — | — |
| **g4 f4** | 1.00 | — | — | — | — | — | — | — |
| **a4 f4** | — | — | 0.50 | — | 0.50 | — | — | — |
| **bf4 a4** | — | — | — | 1.00 | — | — | — | — |
| **bf4 bf4** | — | — | — | — | — | 1.00 | — | — |
| **c5 a4** | — | — | — | 1.00 | — | — | — | — |

Interaction 19-13. Listening to second order Happy Birthday.

```
cm> (events (play-bday 2 40 .15) "bday.midi")
"bday-3.midi"
cm>
```

- bday-3.midi

### Happy Birthday, orders 3-5

```
(markov-analyze bday :order 3 :print? :table :pattern? false)
(markov-analyze bday :order 4 :print? :table :pattern? false)
(markov-analyze bday :order 5 :print? :table :pattern? false)
```

Interaction 19-14. Listening to Happy Birthday, orders 3, 4 and 5.

```
cm> (events (play-bday 3 40 .15) "bday.midi")
"bday-3.midi"
cm> (events (play-bday 4 40 .15) "bday.midi")
"bday-4.midi"
cm> (events (play-bday 5 40 .15) "bday.midi")
"bday-5.midi"
cm>
```

- bday-3.midi
- bday-4.midi
- bday-5.midi

## Amazing Grace

As a final example of Markov processes I offer a piece I composed from a Markov analysis of the spiritual Amazing Grace. The piece also uses weighted random selection and envelopes, both of which have been covered in Chapters 15 and 17. The only real difference between the Markov processes you will hear in the composition and the process described in this chapter is in the manner in which probability weights were controlled in the table. Rather than develop multiple tables to represent the folk song I implemented a single large table whose probability weights were expressed as envelopes. The probability in this large Markov table was then controlled as a function of time.

Amazing Grace is an algorithmic fantasy for the Yamaha Disklavier (a MIDI controlled piano) based on the American folk song of the same name. The fantasy involves a processes of becoming, in which melodic and rhythmic contours of the folk song serve as gravitational centers for the musical texture to coalesce around, like dust spiraling inward to form a star. Amazing Grace begins in a mode and texture very distant from the folk melody. As the composition unfolds the original tune gradually exerts more and more influence over the stochastic texture. Short melodic motives and rhythmic figures first appear and are followed by progressively longer gestures and melodic contours coming to the fore. The process continues until the point of maximum influence, at which time the fantasy has congealed into a block texture with a single rhythmic motive and melodic content completely determined by the folk song. Amazing Grace was commissioned for the opening ceremony of the Multimedial III Festival in Karlsruhe Germany, where it received its world premiere in 1995.

- Amazing-Grace-KA.mp3 stereo MP3, 8 Mbytes, 7 minutes.

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file markov.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 20 Patterns and Composition

Art is the imposing of a pattern on experience, and our aesthetic enjoyment is recognition of the pattern.

— Alfred North Whitehead

It is hard to conceive of a musical work that does not express patterns somehow. Patterns can operate at all levels of a composition, from the transitory surface flow of notes and rhythms to its deep structural organization. A pattern might involve a single sound parameter such as frequency, or it may involve multiple sound attributes evolving in parallel over time. Because patterns permeate musical expression, inventing and manipulating them is fundamental to the work that a composer does. In this chapter we explore an object oriented language for musical pattern description based on a class of object called **`pattern`**. Every subclass of pattern organizes and generates data according to class specific ordering rules. The pattern language provides a number of different subclasses to work with. These patterns can be used individually or combined together to form *composite patterns* that exhibit the characteristics of several different pattern types simultaneously. Table 20-1 contains a summary of the basic pattern types that will be discussed in this chapter. Be sure to consult the Common Music Dictionary for full details about each pattern type as it is introduced over the course of this chapter.

Table 20-1. Table of pattern classes.

| class | description |
|---|---|
| **`cycle`** | sequences elements in a continual loop. |
| **`line`** | sequences elements and sticks on last. |
| **`palindrome`** | sequences elements forwards and backwards. |
| **`heap`** | permutes elements randomly. |
| **`random`** | selects elements in weighted distribution. |
| **`rotation`** | permutes elements by rotation rules. |
| **`markov`** | generates elements by markov process. |
| **`graph`** | generates elements by state transition rules. |
| **`funcall`** | computes elements by calling a function. |
| **`rewrite`** | rewrites elements using grammar rules. |

The pattern language also provides some pattern classes that implement specialized functionality. These specialized patterns (Table 2) are almost always used in conjunction with the general patterns listed in Table 1. Their use will be discussed at appropriate places over the course of this chapter.

Table 20-2. Table of specialized pattern classes.

| class | description |
|---|---|
| **`join`** | forms a single pattern out of multiple patterns. |
| **`copier`** | repeats periods of a pattern. |
| **`chord`** | produces lists of elements. |
| **`transposer`** | transposes patterns and lists. |
| **`range`** | generates numerical ranges. |
| **`pval`** | evaluates a Lisp expression embedded in a pattern. |

Before examining the pattern classes in more detail we begin first by describing characteristics that all patterns share and then demonstrating how patterns are created and used to organize musical data.

## Pattern Properties

The manner in which a pattern generates data is influenced not only by what specific pattern type the class implements but also by a set of common properties that influence the way in which all patterns behave. One of the most important properties of a pattern is its *period length*. All patterns group their data into "chunks" called periods. The period length of a pattern determines the number of elements that constitute a chunk's worth of data. The *repeat* property of a pattern places an optional ceiling on the number of periods, or chunks, that the pattern will be allowed to produce. If a pattern reaches its repeat limit it will only generate an *end of data* marker from that point onward. This marker allows a caller to determine when a pattern is done producing elements without knowing in advance how many elements a pattern will produce. A pattern's *name* property can be used to reference a pattern repeatedly as a "motive", or building-block, inside the definition of a surrounding pattern. Patterns can contain any type of Lisp data, including *subpatterns*. A subpattern defines a local ordering of elements within the surrounding pattern. As a general rule, any value specified to a pattern can be replaced by a pattern of values. This rule applies not only to pattern data, but to many of the pattern's properties as well. In this way the characteristics of a pattern can change and evolve along with the data that the pattern generates.

## Working with Patterns

Patterns are created using the **`new`** macro. All patterns support the following property initializations:

:of *data*
: Sets the pattern's elements to *data*. This initialization can also be specified using the **`:notes`** or **`:keynums`** aliases,in which case *data* is parsed as a list of note names or key numbers, respectively.

:name {*string* | *symbol*}
: Sets the name of the pattern to *string* or to the print name of *symbol*.

:for {*number* | *pattern*}
: Sets the period length of the pattern to *number* or *pattern* of numbers.

:repeat {*number* | *pattern*}
: Sets the repetition factor of the pattern to *number* or *pattern* of numbers.

:counting {:periods | :values}
: Determines how the pattern counts subpatterns. The default value is **`:periods`**, which means each period of a subpattern is counted as a single choice in the pattern.

Example 20-1 demonstrates three simple pattern definitions.

Example 20-1. Pattern creation using the new macro.

```
(new cycle :of '(a b c))

(new heap :keynums '(c4 b d5 ef3) :for 20
     :name 'gesture1)

(new heap
  :of (list (new cycle :of 'q :for 3)
            (new palindrome :of '(e. s e))
            (new random :of '(h w h.) :for 1)))
```

The first expression in Example 20-1 creates a cyclic pattern of three symbols: a, b and c. The second expression defines a random permutation of key numbers with twenty key numbers per period. The third expression creates a composite rhythmic pattern built from three different subpatterns.

## Reading Pattern Values

Once a pattern has been created its elements are accessed, or *read*, using the **`next`** function:

[Function]
(**next** *pattern* [*amount*])

The **`next`** function returns the next value, or list of values, from *pattern*. The optional *amount* parameter determines how many values are read from the pattern. If *amount* is not specified then just the next element in the pattern is returned. If *amount* is an integer then that number of values are read and returned by the pattern as a list. If *amount* is not an integer and it is not false then a period's worth of elements is read from the pattern and returned in a list.

Interaction 20-1. Creating and reading patterns.

```
cm> (define p (new cycle :of '(a b c)))

cm> (next p)
a
cm> (next p)
b
cm> (next p)
c
cm> (next p #t)
(a b c)
cm> (next p 7)
(a b c a b c a)
cm> (next p #t)
(b c)
cm>
```

Example 20-1 creates a cyclic pattern and reads it using the **`next`** function. A cycle generates its elements in sequential order and then loops back to its first element after the last value has been produced. Since no **`:for`** initialization was specified when the pattern in Example 20-1 was created its period length was automatically set to three, the number of elements in the pattern. The fourth call to **`next`** specifies true for the *amount* parameter, which causes one full period of the pattern elements to be read and returned in a list. Note that the last expression in Example 20-1 only returns two value in the period because the preceding expression read seven values and so left only two values remaining in the current period.

## Subpatterns

Patterns can contain other patterns, either as data or to exert control over the pattern's generative properties. Each internal pattern, or *subpattern*, controls a local ordering within the larger pattern's scope. Subpatterns are read for their entire period before any other elements in the surrounding data is considered. Normally all of these elements count as a single selection in the period count of the larger pattern. A subpattern in a pattern's property produces a new value each time the surrounding pattern starts a new period. We will use a simple first example to demonstrate some of the basic features of subpatterns.

## Figures and Motives

Every composer is familiar with the *Alberti Bass*, a stereotypical "broken chord" figure that appears in many works from the Common Practice period. Example 20-2 shows the definition of two Alberti figures using the cycle pattern. Each cycle defines a four-note version of the Alberti bass in which the second and fourth tones repeat the same note.

Example 20-2. Alberti figures implemented as patterns.

```
(new cycle :notes '(c3 e g e) :name 'm1)

(new cycle :notes '(d3 f g f) :name 'm2)
```

The figures were given the names m1 and m2 so that they can be easily referenced as motivic units inside a larger pattern. Naming is a convenience that allows a pattern to be fetched using the #& feature without having to resort to global variables to store the pattern object. In Example 20-3 the #&m1 and #&m2 patterns are used as building blocks to form a little bass line in which each subpattern appears several times. Since a cycle continually loops its data from back to front, the Alberti figures in Example 20-2 will begin with their first note each time they are repeated in the surrounding pattern.

Example 20-3. Alberti figures used in a short Alberti bass line.

```
(new cycle :name 'bass-line
     :of (list #&m1 #&m1 #&m2 #&m2
               #&m1 #&m2 'c3))
```

- pat-1.midi

## Multidimensional Patterns

The short bass line pattern shown in Example 20-3 generates values for a single sound parameter. But musical motives and gestures often involve characteristics from more than one dimension of sound linked together as a single, aggregate unit. For example, the opening motive of Beethoven's Fifth symphony is as much, or more, a rhythmic motive as it is a pitch motive. In simple cases, motives that involve multiple sound parameters can be manipulated using separate patterns to control each sound parameter independently but in parallel. However, parallelism neither reflects the underlying conceptual unity of the motive nor "scales" well as the size and complexity of the motivic patterns increases. Worse yet, if the motivic patterns are not completely deterministic, or are not all of the same pattern type, then parallel patterns will not generate values in the required synchronized order.

The pattern language provides a special type of pattern called **`join`** which can be used to link more than one pattern together forming single units of structure that will always produce synchronized values regardless of how the joined patterns are organized inside a larger pattern. Example 20-4 rewrites the bass line from the preceding example in terms of joins that link each Alberti pitch pattern with a unique rhythmic pattern to form two-dimensional musical motives.

Example 20-4. Frequency and rhythm joined together.

```
(new cycle
  :of (list (new join :name 'j1
                 :of (list #&m1
                           (new cycle :of '(te ts te ts))))
```

```
         #&j1
         (new join :name 'j2 :of (list #&m2 's))
         #&j2 #&j1 #&j2 '(c3 h)))
```

- pat-2.midi

In the first join every note from m1 combines with a corresponding triplet rhythm from a cyclic rhythm pattern. The second join links each note in m2 with a sixteenth note value. The very last element in the pattern **(c3 h)** defines a half-note C3 value.

Joined motives with random characteristics still maintain their conceptual integrity as single units of structure. Example 20-5 demonstrates a pattern that joins frequency, rhythm and amplitude values to form gestures involving three dimensions of sound. The overall pattern is a cycle of three local gestures, the second and third gesture produce random variations each time they are generated from the larger pattern.

Example 20-5. Three dimensional motives with random variation.

```
(new cycle
  :of (list (new join
              :of (list (new cycle :of '(ef5 e4 r))
                        (new cycle :of '(s s q) )
                        'ff))
            (new join
              :of (list (new heap :of '(g4 fs4 fn5 b5))
                        (new cycle
                          :of (list (new heap :of '(e q e))
                                    'h))
                        'p))
            (new join
              :of (list (new cycle :of '(af5 f5 a4 cs4 c5))
                        (new heap :of '(e s q) :for 5)
                        (new cycle :of '(pp p mp mf f))))))
```

- pat-3.midi

## Composite Patterns

*Composite patterns* are patterns that exhibit the characteristics of more than one type of pattern at the same time. Example 20-6 presents a simple example of combining cyclic and random behaviors together. The outer **cycle** pattern sequences three **heap** subpatterns. The heaps in Example 20-6 collectively partition a single octave into three groups of chromatic notes, where each group lies a major third above the previous group. A **heap** pattern behaves like a deck of cards: it shuffles its elements each time it starts over. Since the heaps in Example 20-6 randomly permutated their elements the series of values that results from the composite pattern will exhibit localized random orderings within a larger periodic ascent.

Example 20-6. Cycles of random permutation.

```
(new cycle
  :of (list (new heap :notes '(c3 cs d ef))
            (new heap :notes '(e3 f fs g))
            (new heap :notes '(gs3 a as b))))
```

- pat-4.midi

If the pattern in Example 20-6 is turned "inside out" then randomly ordered cycles will result:

Example 20-7. Randomly permuted cycles.

```
(new heap
  :of (list (new cycle :notes '(c3 cs d ef))
            (new cycle :notes '(e3 f fs g))
            (new cycle :notes '(gs3 a as b))))
```

- pat-5.midi

## Pattern Periods

The period length of a subpattern has a large impact on how the overall pattern evolves. If, as in the preceding examples, the period length is the same as the number of elements then all the elements of a subpattern will be generated before any other data in the surrounding pattern will be considered. This sort of traversal is called *depth first ordering*. By setting the period length of a subpattern to less then the number of its elements, the subpattern will yield to other patterns in the surrounding data and produce *breadth first* ordering.

We can observe the effect of a breadth-first traversal by setting the period lengths of the of inner heaps in Example 20-6 to 1. A period length of 1 will cause each heap to produce only 1 note per period of the outer cycle. This will generate a different four notes from the three heaps each period of the outer cycle, thus turning the pattern into a "random generator" of Alberti-like figures.

Example 20-8. Randomized Alberti figures.

```
(new cycle
  :of (list (new heap :notes '(c3 cs d ef) :for 1)
            (new heap :notes '(e3 f fs g) :for 1 :name 'm)
            (new heap :notes '(gs3 a as b) :for 1)
            #&m))
```

- pat-6.midi

Note that the fourth element #&m in the pattern does not produce a repeated value in the melodic output because #&m is a repeated pattern, not a repeated value. To make our generator repeat the exact same value for the second and fourth element will require that the fourth element literally reference whatever value the middle pattern #&m last produced. The function **`pattern-value`** can be used to return the last value that a pattern generated:

Interaction 20-2. Referencing the last value a pattern produced.

```
cm> (define p (new cycle :of '(a b c)))

cm> (pattern-value p)
:not-a-datum
cm> (next p)
a
cm> (pattern-value p)
a
cm>
```

The first call to **`pattern-value`** returned the special keyword token **`:not-a-datum`** because the pattern had no value at that point. Once **`next`** reads a value then **`pattern-value`** can fetch that value again from the pattern that produced it.

Note that in the case of our random Alberti generator the **`pattern-value`** expression will have to be evaluated *inside* the pattern, every time the fourth element is to be read. This means that the expression **`(pattern-value #&m)`** must appear in the list of data specified to the pattern when it is created. But this presents problem: the list of data specified to the new macro is evaluated when the pattern is *created*, not when the pattern is read. This means our call to pattern-value must somehow be *delayed* until the pattern is actually read using **`next`**. A special pattern macro called **`pval`** (pattern value) can be used to delay the evaluation of a Lisp expression until it is encountered by the **`next`** function in a pattern.

[Macro]

(**`pval`** *expr*)

Pval delays the evaluation of a Lisp expression *expr* until the pval is encountered in a pattern by **`next`**. Pval allows any Lisp expression to be placed and evaluated in a pattern. A pval expression will be re-evaluated every time it is encountered by **`next`** in the pattern.

Example 20-9. Using pval to delay evaluation of an expression.

```
(new cycle
  :of (list (new heap :notes '(c3 cs d ef) :for 1)
            (new heap :notes '(e3 f fs g) :for 1 :name 'm)
            (new heap :notes '(gs3 a as b) :for 1)
            (pval (pattern-value #&m))))
```

- pat-7.midi

The pval in Example 20-9 delays the evaluation of the call to **`pattern-value`** until the pattern is read using the **`next`** function. When the fourth element is accessed the pval returns the value of its expression in its place. This expression will be re-evaluated each time the fourth element is accessed, effectively repeating whatever value was previously produced by the #&m subpattern.

As a final variation on the Alberti bass idea we will place our randomized Alberti generator inside a **`copier`** pattern. A **`copier`** is a specialized pattern that generates a period from a source pattern and then repeats, or copies, that period some number of times before generating the next new period of data from the source. The period length of the copier is set to a heap that produces either two, three or five iterations of each period of the random figure.

Example 20-10. Using a copier to repeat pattern periods.

```
(new copier
  :of
  (new cycle
    :of (list (new heap :notes '(c3 cs d ef) :for 1)
              (new heap :notes '(e3 f fs g) :for 1 :name 'm)
              (new heap :notes '(gs3 a as b) :for 1)
              (pval (pattern-value #&m))))
  :for (new heap :of '(2 3 5)))
```

- pat-8.midi

### Zero-length Periods

A subpattern that sets its period length to zero "disappears" from the surrounding pattern until it is reselected with a period length greater than zero. In the following example, the note B4 will periodically appear three, two and then zero times in the surrounding pattern.

Example 20-11. Subpatterns with zero period length are skipped over.

```
(new cycle :of `(c4
                 ,(new cycle :of 'ef4
                       :for (new cycle :of '(1 2)))
                 ,(new cycle :of 'b4
                       :for (new cycle :of '(3 2 0)))
                 d4))
```

- pat-9.midi

Example 20-11 also demonstrates a very useful Lisp feature called the *backquoted list*. The pattern's data list contains two note names (C4 and D4) that are tokens for the pattern to generate and two expressions that, when evaluated, create subpatterns. Recall from Chapter 5 that a backquoted list acts just like a quoted list except that expressions preceded by a comma "," are explicitly evaluated. In the case of pattern data, the backquote is useful for notating data lists in which some elements are pattern tokens while others elements are *expressions* that create subpatterns.

## Patterns and Processes

Patterns can greatly simply the overall design and implementation of a musical process. For example, most of the code in the first processes definition in Chapter 14 (Piano Phase) was specifically related to treating lists in a cyclic manner. Recasting the process to use a cycle pattern to hold the phasing trope eliminates the need for several local variables and makes the code more compact and easier to understand.

Example 20-12. Process from Chapter 14 converted to use a pattern.

```
(define (piano1 trope amp chan)
  (let* ((cycl (new cycle :keynums trope
                      :repeat (length trope)))
         (rate (rhythm->seconds pp-pulse pp-tempo)))
    (process for k = (next cycl)
             until (eod? k)
             output (new midi :time (now)
                         :keynum k
                         :duration (* rate 1.5)
                         :amplitude amp
                         :channel chan)
             wait rate)))
```

Example 20-12 demonstrates the basic manner in which patterns are incorporated into a process definition. Data for the pattern is usually passed to the process as a list and the process creates a local pattern object to hold the data. At each process iteration a **`for`** stepping variable reads successive values from the pattern. In Example 20-12 the value of *k* is set to a pattern value which is then assigned to the keynum of each MIDI object output to the score.

Creating patterns inside a process is almost always preferable to referencing patterns defined globally in the Lisp environment. There are several reasons for this. First, a pattern created inside a process is guaranteed to start from its initial conditions each time the process function is called. In contrast to this, a pattern that is defined globally will resume from the point it last stopped unless the composer remembers to redefine the global pattern before it is used again. Second, if the composer uses a local process variable to hold the pattern then no other process can share that exact same pattern object. If a single pattern is read by multiple processes then each process will read alternate values from the pattern rather than successive values. Sharing a pattern between processes can be useful but in most cases it is probably not what a composer intends.

## Pattern State

The current *state* of a pattern can be just as important to a process as the data that the pattern produces. To illustrate, the **`:repeat`** initialization in Example 20-12 established a limit on the number of periods the cyclic trope produced. A process can use this information to determine when to stop generating events rather than by keeping count of its interations. The pattern facility provides several functions that allow pattern states to be "inspected" and tested for certain conditions.

[Function]
    (**eop?** *pattern*)
[Function]
    (**eod?** *pattern*)
[Function]
    (**pattern-value** *pattern*)
[Function]
    (**pattern-period-length** *pattern*)

The **`eop?`** (end of period) function returns true if *pattern* is currently at the end of a period, and the **`eod?`** (end of data) predicate returns true if a pattern or a pattern value has reached an established **`:repeat`** limit. The **`pattern-value`** function returns the previous value read from a pattern (if any) and **`pattern-period-length`** returns the current period length of the pattern if one has been established. Example 20-13 demonstrates how these functions might be used to control the actions of a process.

Example 20-13. Using pattern states to control a process.

```
(define (play-pat reps rate dur amp )
  (let ((pat (new heap
               :notes '(c4 d ef f g af bf c5)
               :for (new random :of '(8 12 16))
               :repeat reps))
        (x 0)
        (r #t ))
    (process for k = (next pat)
             until (eod? k)
             for p = (pattern-period-length pat)
             if r set x = 0
             output
             (new midi :time (now)
                  :keynum k
                  :amplitude (interp x 0 .4 (- p 1) amp)
                  :duration dur)
             wait rate
             set r = (eop? pat)
             set x = (+ x 1))))
```

The **play-pat** process creates a heap pattern to generate *reps* periods of notes. Since the heap was given a random period length the process cannot know in advance how many events it will generate. A test using the **eod?** function allows the process to determine that the pattern's end-of-data marker has appeared once all of the events in *reps* periods have been produced. The **until** clause is used to kill the process as soon as that point has been reached.

The process in Example 20-13 produces a crescendo over every period in the heap pattern without knowing in advance how long each period actually is. The amplitudes of events within a single period are calculated using linear interpolation (Chapter 16 and range from .4 to the maximum *amp* value specified to the process. The variable *x* provides the interpolation lookup value, and ranges from 0 to one less than *p*, the heap's current period length as determined by the **pattern-period-length** function. The variable *r* is a flag that marks the beginning of each new period. When *r* is true then the *x* lookup variable is reset to 0 to begin a new crescendo. The variable *r* is updated at the end of each iteration by the **eop?** function, which returns true only if the pattern is at the end of its current period. Note that **eod?** testing will typically occur at the beginning of a process definition, before dependant operations actually reference pattern values, and **eop?** testing will usually occur at the end of the process definition, after a value has been read and processed, because that value may have been the last value in the current period.

Interaction 20-3. Listening to a pattern.

```
cm> (events (play-pat 10 .1 .3 .8) "pat-10.midi")
"pat-10.midi"
cm>
```

- pat-10.midi

## Pattern Sharing

If multiple processes share pattern data then each process can output its own stream of musical events in which some sound properties depend on a common shared "idea" while others remain local to each process definition. One simple way to implement a shared idea is to treat one of the processes as a "master" that sets up and manages the shared data and to treat the other processes as "slaves" that reference shared data but do ever set it. Care must be taken to insure that the master process executes ahead of the slave processes so that the shared data is ready before the slaves start using it. This can be done either by scheduling the master process slightly before the slaves, or by insuring that the master process is the first process in the list of processes passed to the events function.

Example 20-14 defines two processes that harmonize a random bass line using a shared pattern value *the-note*. The first process is the "master" process, it reads a value from its random bass line pattern, outputs a midi event to play the note, and sets the shared variable *the-note* to that pattern value. The second process is a "slave" process that generates a chord on whatever bass note the master process has last produced. The slave process sets *c* to a randomly selected chord from a pattern of chord templates and then transposes each chord two octaves above the current value of *the-note*. The slave's **each** clause is used to iterate over each value in the current chord list to output a midi event.

Example 20-14. Sharing pattern values between processes.

```
(define (two-hands end tempo)
  (let ((the-note #f))  ; shared pattern value
    (list
     ;; master process creates the bass-line and sets
     ;; the shared variable.
     (let ((bass (new random
                   :keynums '(e1 f g a b c2 d)))
           (rhys (new random :of '( h.. h e))))
       (process while (< (now) end)
                for k = (next bass)
                for r = (rhythm (next rhys) tempo)
                set the-note = k
                output (new midi :time (now)
                            :keynum k :duration r
                            :amplitude .8)
                wait r))
     ;; slave procees adds random harmony two
     ;; octaves above the-note
     (let ((rhys (new random :of '((q max 3) (e max 2)
                                    (s max 1)
                                   (t weight .5 max 1))))
           (harm (new heap :of '((0 3 7 8) (1 5 8 10)
                                  (0 3 7 10) (1 5 8 12)
                                  (1 3 7 10) (3 5 8 12)
                                 (1 5 7 10)))))
       (process while (< (now) end)
                for c = (transpose (next harm)
                                   (+ the-note 24))
                for r = (rhythm (next rhys) tempo)
                each k in c
                output (new midi :time (now)
                            :keynum k :duration (* r 2)
                            :amplitude .4)
                wait r)))))
```

- pat-11.midi

## Pattern Classes

Most of the patterns we have seen so far have very simple control features and are relatively straightforward to use. But note that the slave process in Example 20-14 contains the definition of a **`random`** pattern where each element in the pattern has been specified as a list containing the pattern element followed by certain pattern *node properties*, such as a **`max`** value or a **`weight`** factor. In this section we examine pattern classes that, like the **`random`** pattern, involve either special control features, special formats in their data specification, or both.

### Palindrome

A **`palindrome`** pattern enumerates its data forwards and backwards. Subpatterns are visited in palindromic order but will continue to produce their own local orderings within the palindrome. The palindrome's **`:elide`** initialization controls how the beginning and ending elements in the pattern data are treated when the palindrome reverses direction. Table 1 demonstrates the effect of the all the possible :elide values given example data of four letters: A B C D.

Table 20-3. Effects of :elide on palindrome data.

| **elide value** | **result** |
|---|---|
| false | A B C D D C B A \| A B ... |
| true | A B C D C B \| A B ... |
| **`:first`** | A B C D D C B \| A B ... |
| **`:last`** | A B C D C B A \| A B ... |

Example 20-15. A palindrome with subpattern.

```
(new palindrome
  :notes `(, (new heap :notes '(c4 b3 bf a af g) :for 1)
             d4 ef f gf)
  :elide (new random :of '(:first #f)))
```

- pat-12.midi

The palindrome in Example 20-15 contains 5 elements, the first of which is a heap subpattern with a period length of 1. The **`:elide`** value of the palindrome is set to choose between **`:first`**, which causes the heap pattern to not repeat as the last element in the palindrome, and false, which causes it to repeat.

## Rotation

A **`rotation`** pattern permutes its elements according to rules specified in the **`:rotations`** initialization. The value of **`:rotations`** should be a *rotation list* or a pattern that generates rotation lists. A rotation list contains (up to) four numbers that describe which elements in the pattern will be "swapped", or exchanged, to produce the next rotation. This swapping is applied iteratively over all the data to produce the rotation for the next period. The format of a rotations list is:

(*start step width end*)

where *start* is the initial index (zero based) in the data to begin swapping, *step* is the distance to move in the data after each swap, *width* is the distance between swapped elements, and *end* marks the index in data to stop element swapping. If the **`:rotations`** initialization is not provided it defaults to (0 1 1 *end*), which causes the front element to shift to the back on each new period. For example, if **`a b c d`** represents the initial order for a rotation, then the default rule (0 1 1) iterates over all elements and swaps adjacent pairs, thus moving the first element **`a`** to its next position at the end of the list:

Table 20-4. Rotation of front element A to back.

| | |
|---|---|
| **initial** | **a** b c d |
| **swap first pair** | b **a** c d |
| **swap second pair** | b c **a** d |
| **swap last pair** | b c d **a** |

The rotation in Example 20-16 consists of five elements, the first of which is a subpattern.

Example 20-16. The rotation pattern.

```
(new rotation
  :notes `(, (new range :by -2 :initially (keynum 'c6))
             b3 f4 g5 ef6)
  :rotations (new cycle :of '((0 1 1) (0 1 2))))
```

- pat-13.midi

The first element in the rotation is a specialized pattern called a **`range`** that can generate either bounded or unbounded number ranges. The range used in Example 20-16 returns a series of descending key numbers a whole step apart starting from key number 84 (C6). The **`:rotations`** for the pattern is expressed as a cycle of of two rules so that the pattern alternates rules on each period of the pattern. The first rule is the default rotation (0 1 1) and the second swaps elements two steps apart, i.e. elements separated by one element between them.

The example file cring.cm contains some examples of rotations that implement *change ringing* patterns, complex rotational schemes developed by English church bell ringers for generating long sequences of peals without direct repetition.

## Random

The **random** pattern generates elements in a discrete weighted distribution. By default, each element in a random pattern has an equal probability of selection and may be repeated in direct succession any number of times. This default behavior can be modified by specifying the element as a *node*, a list containing the element followed by a set of optional keyword constraints to place on the element:

```
(x [:weight n] [:min i] [:max i])
```

*X* is the pattern element, **:weight** sets the probability of the node relative to other elements in the pattern, **:min** sets a lower bound on how many direct repetitions of the node element must be made before a new element might be selected, and **:max** sets a ceiling on how many direct repetitions of the node element may be made before a new element must be selected. An element's **:weight** value may be specified as a *pattern* of weights, in which case a new weight will be selected for each period of the random pattern. Note that the min and max node values will affect the statistical accuracy of the node's probability weight.The process is Example 20-17 uses the random pattern to generate transposition offsets, rhythmic patterns, and amplitude values.

Example 20-17. Random 12-tone generator.

```
(define (ranrow end)
  (let ((rows (new transposer
                :of (new cycle :of '(0 1 6 -5 10 11 5 16 15 21 26 20))
                :on (new random :keynums '(c4 cs3 b3 fs4 fs3))))
        (rhys (new random
                :of `((t :min 4) (s :min 2)
                      (q :weight 3)
                      (te :min 3) (tq :min 3)
                      ,(new cycle :of '(e. s))
                      (e :weight 2))))
        (amps (new random :of '((mf :weight 7)
                                (mp :weight 3)
                                p))))
    (process while (< (now #t) end)
             for k = (next rows)
             for r = (rhythm (next rhys) 70)
             for a = (amplitude (next amps))
             output (new midi :time (now)
                         :keynum k :amplitude a
                         :duration r)
             wait r )))
```

In this example the twelve-tone row is specified as a cycle pattern that is transposed to key numbers selected from a random pattern. (The intervals in the row are not notated within a single octave to provide a slight contour to the melodic line.) A **transposer** is a specialized pattern that can transpose either patterns or lists of key numbers or notes. The rhythmic values are also selected at random; the **:min** feature is used to insure that rhythmic values of less than an eighth note duration are reselected enough times so as to place the next different rhythm at the beginning of an eight-note pulse.

- pat-14.midi One channel, piano.

## Markov

The **`markov`** pattern implements nth order Markov chains. The data specified to the pattern is a list of *transition rules* where each rule is a list of the form:

```
(past... -> next ...)
```

where *past* is zero or more identifiers that define the transition trigger of the rule and values to the right of the marker **`->`** define the transition's outcomes. The number of ids the left-hand side contains determine the Markov order of the pattern and their left to right order define the succession from most distant past to most recent choices. The special id **`*`** is a "wild card" that can appear in the left hand side to match any past id at that position. This allows higher order tables to "collapse" transitions with identical outcomes and near-identical left-hand sides into a single transition rule.

By default each transition outcome has the same probability of being selected. To alter the weight of an particular outcome relative to the other outcomes, specify its id in a list together with its weight. For example the transition rule:

```
(c -> a (b 2) c)
```

defines a 1st order Markov transition whose outcome **`b`** has twice the selection probability than either the **`a`** or **`c`** outcomes. The transition rule:

```
(q * a -> e r (d 3) g)
```

describes a third-order transition triggered whenever **`a`** is the most recent past outcome and **`q`** is the third past outcome. Example 20-18 shows a second order markov pattern that generates rhythms with an underlying pulse. Using the wildcard token * reduces the number of transition rules that would otherwise have to be entered.

Example 20-18. A second order Markov rhythm pattern.

```
(new markov
  :of '((e s -> s)
        (q s -> e.)
        (q. s -> s)
        (h s -> e.)
        (e e -> s e e. q q. q.. h)
        (* e -> e)
        (* s -> e e. q q. q.. h)
        (* h -> e e. q q. q.. h)
        (* q.. -> s)
        (* q. -> e s)
        (* q -> e e. q q. q.. h)
        (* e. -> s))
  :for 20
  :past '(q q))
```

The optional **`:past`** initialization for markov patterns "preloads" the pattern's memory buffer with an explicit starting condition for the process. The value **`(q q)`** will cause the second to last transition rule in the pattern to be triggered when the pattern is read.

- pat-15.midi Drum Track (Ride Cymbal)

The example files foster.cm and foster-1.midi contain a large example of a second order Markov process that generates melodic material based on the analysis of Stephen Foster's folk songs described in chapter eight of Computer Music by Charles Dodge and Thomas Jerse.

## Graph

A graph pattern generates elements from *transition rules* associated with the elements in the pattern. Elements and their transtions are represented as *graph nodes*. A graph node is a list in the form:

```
(x [:id n] [:to ids])
```

where *x* is the element in the graph node, **`:id`** sets the node's *identifier*, or name, and **`:to`** is its transition rule. The transition rule determines the next node to select if the current node is selected. The value of can be a node identifier or pattern of identifiers. Note also that if the **`:to`** value of each node is a **`random`** pattern then the graph will implement a first order Markov process. The **`:to`** initialization can also be specified as "right arrow": **`->`**. (The right arrow in a node description stands for "produces".) The graph in Example 20-19 generates a melodic line using both random and cyclic transition rules. The graph is placed inside a cyclic pattern that alternates one period of the chant with a rest. The graph itself contains 5 elements. Each element is represented by a graph node with an identifier that uniquely "names" that element in the graph, and a transition rule that will produce the next node if that node is selected. In Example 20-19 the transition rules are expressed as subpatterns that produce the identifiers of the successor nodes. The first four nodes in the graph select their successor nodes by randomly choosing between two possible successors (first order Markov). The fifth node selects its successor by cycling through the first four nodes. The pattern element of the fifth node is itself a heap subpattern that produces a four note melodic fragment each time it is selected. The entire graph is embedded in a cyclic pattern that plays one period of the graph followed by a rest.

Example 20-19. A quasi-markov graph

```
(new cycle
  :of `(,(new graph
           :of `((c4 :id 1 :to ,(new random :of '(2 5)))
                  (d4 :id 2 :to ,(new random :of '(1 3)))
                  (ef4 :id 3 :to ,(new random :of '(2 4)))
                  (f4 :id 4 :to ,(new random :of '(3 5)))
                  (,(new heap :of '(g4 fs4 b4 c5))
                   :id 5
                   :to ,(new cycle :of '(1 2 3 4)))))
        r))
```

- pat-16.midi

## Funcall

The **`funcall`** pattern implements patterns by function call. Each time a funcall pattern requires a new period of elements it calls a user specified function to return a list of value that become the next period of values for the pattern. The best way to work with the funcall pattern is to define functions that create and return them. That way the function can be called with different arguments to create different versions of the funcall pattern. This is demonstrated in the next example which creates a funcall pattern to return random values between a low and high bounds without any direct repetition.

Example 20-20. Implementing a pattern with the funcall pattern.

```
(define (noreps hi low n)
  (define (thunk )
    (loop repeat n
          for x = (between hi low x)
          collect x))
  (new funcall :of #'thunk))
```

The **`noreps`** function returns a funcall pattern that will generate *n* random values per period between the *hi* and *low* values without direct reselection. **`noreps`** defines a helper function, called **`thunk`**, to do most of its work. The term *thunk* is used by Lisp programmers to describe a function that is called without any arguments to produce an effect. The thunk for **`noreps`** uses **`between`** to generate a list of random values without any direct repetitions. This thunk is then passed to the funcall pattern as the function to call whenever the pattern needs a new period of values. Interaction 20-4 demonstrates some sample output from a **`noreps`** pattern.

Interaction 20-4. Defining a noreps pattern.

```
cm> (define p (noreps 60 72 5))

cm> (next p t)
(63 61 64 62 70)
cm> (next p t)
(66 61 67 69 61)
cm>
```

We can use **`noreps`** in conjunction with the **`copier`** pattern to create a wandering melody in which locally random figures are repeated to produce larger scale periodicity.

Example 20-21. Repetition and randomness.

```
(define (wander tmpo)
  (let ((bass (new copier :of (noreps 41 60 5)
                   :for (new random :of '(1 3 5))
                   :repeat 10))
        (amps (new heap :of '(.6 .4 .4 .4 .4))))
    (process for k = (next bass)
             for a = (next amps)
             until (eod? k)
             output
             (new midi :time (now)
                  :keynum k
                  :duration (if (> a .5) 5 3)
                  :amplitude a)
             wait (rhythm 1/20 tmpo))))
```

The **`wander`** process creates a melody that selects different random fragments from a range and then plays each fragments one, three or five time before selecting a new one. The amplitude pattern for the process accents exactly one note during each fragment so that an irregularly spaced melodic line is "brought out" from the steady background of notes in each fragment.

Interaction 20-5. Listening to wander.

```
cm> (events (wander 48) "pat-17.midi")
"pat-17.midi"
cm>
```

- pat-17.midi One channel, monotimbral.

In this next example we use **`funcall`** to implement three new patterns. The **`revpat`** function creates a funcall that reverses (retrogades) a source pattern. The **`invpat`** pattern will invert a pattern and **`mirror`** will produce a strict palindrome of a pattern.

Example 20-22. Retrograding and inverting patterns.

```
(define (revpat pat)
  ;; return the reverse of pat
  (define (thunk )
    (let ((vals (next pat true)))
      (reverse vals)))
  (new funcall :of #'thunk))

(define (invpat pat)
  ;; return the inversion of pat,
  ;; which must contain keynums or notes.
  (define (thunk )
    (let ((vals (next pat true)))
      (invert vals)))
  (new funcall :of #'thunk))

(define (mirror pat)
  ;; return a period of pat followed
  ;; by its strict retrograde
  (define (thunk )
    (let ((vals (next pat true)))
      (append vals (reverse vals))))
  (new funcall :of #'thunk))
```

Note, that the retrograding patterns can be applied to any pattern, regardless of how that pattern generates values.

Interaction 20-6. Palindrome of random values.

```
cm> (define p (mirror (noreps 0 12 6)))

cm> (next p t)
(0 2 7 0 2 7 7 2 0 7 2 0)
cm> (next p t)
(4 3 5 10 0 4 4 0 10 5 3 4)
cm>
```

The thunks for **`revpat`**, **`invpat`** and **`mirror`** all function in a similar manner. Each thunk sets a local variable *vals* to the next period of values from the source pattern *pat* and then performs the appropriate operation on that list of data: **`revpat`** reverses the list, **`invpat`** inverts the list (which must therefore contain either key numbers or note names), and **`mirror`** appends the list of values to the reverse of that list. The process in Example 20-23 uses funcall patterns to generate randomly shuffled prime, inversion, retrograde and retrograde-inversion forms of a randomly generated 12-tone row.

Example 20-23. Random rows with retrograde and inversion.

```
(define (ranrow2 end offset tmpo chan)
  ;; get a row, any row.
  (let* ((arow (shuffle
                (loop for i below 12
                      collect i)) )
```

```
        (main (new cycle :of arow)) ; prime form
        (rows (new transposer
                ;;  heap of p, i, r and ri forms.
                :of (new heap
                      :of (list main
                                (invpat main)
                                (revpat main)
                                (revpat (invpat main))))
                :on (new cycle
                      :keynums (transpose arow offset))))
        (rhys (mirror
               (new random
                 :of (list
                      (new cycle :of 't :for 4)
                      (new cycle :of 's :for 2)
                      '(q :weight 3)
                      (new cycle :of 'te :for 3)
                      (new cycle of 'tq :for 3)
                      (new cycle :of '(e. s))
                      '(e :weight 2)))))
        (amps (new random :of '((mf :weight 7)
                                (mp :weight 3)
                                p))))
    (process while (< (now #t) end)
             for k = (next rows)
             for r = (rhythm (next rhys) tmpo)
             for a = (amplitude (next amps))
             output (new midi :time (now)
                          :keynum k
                          :amplitude a
                          :duration r
                          :channel chan)
             wait r )))
```

Each time the **ranrow** process is called it will create a new 12-tone row and use **invpat** and **revpat** to produce the inversion, retrograde and retrograde inversion forms. The *rows* variable contains a heap pattern that randomly permutes the different row forms. Each new row is also used to determine its transposition levels from the key number *offset* specified to the process. The *rhys* variable holds a mirror pattern that produces rhythmic palindromes in which the first half of each period contains a randomly generated rhythmic structure and the second half reverses it. In Interaction 20-7 we listen to four versions of the process, each on a separate channel, where each voice enters 8 seconds later than the last one and all voices stop a the same time.

Interaction 20-7. Random 12-tone music.

```
cm> (events (loop for i below 4
                  collect
                  (ranrow2 32
                          (transpose 'c3 (* i 12))
                          40
                          i))
            "pat-18.midi"
            (loop for i from 0 by 8 to 16
                  collect i))
"pat-18.midi"
cm>
```

- pat-18.midi Four channels.

## Rewrite

The **`rewrite`** pattern produces successive *generations*, of elements, where each generation is created by rewriting the previous generation in term of a grammar defined by the user. A grammar is a set of *rewrite rules*, a set of instructions for "respelling" the elements in successive generations. For example, consider a rewrite pattern whose first generation of data consists of two tokens **`a`** and **`b`** and whose grammar consists of the following two rewrite rules:

> rule 1: **`a -> b a`**
> rule 2: **`b -> a a b`**

The first rule says that every A in the current generation should be replaced (rewritten) with the sequence B A. The second rule says to replace every B in the current generation with the sequence A A B. If the initial generation in the rewrite pattern is set to A B then the rules will produce the following pattern generations:

Table 20-5. Rewrite transformations from rules.

| generation | pattern elements |
|---|---|
| 1 | **`a b`** |
| 2 | **`b a a a b`** |
| 3 | **`a a b b a b a b a a a b`** |
| 4 | **`b a b a a a b a a b b a a a b b a a a b b a b a b a a a b`** |

Generation 2 was created by applying the grammar rules to the entire sequence of generation 1. Rule 1 replaced every **`a`** in generation 1 with **`b a`** and rule 2 replaced every **`b`** with **`b a a`**. This transformed the initial generation **`a b`** into the second generation's sequence **`b a a a b`**. Successive generations are produced in a similar manner, The rewrite rules defined above are additive, which means that successive generations are longer and longer. Rewrite sequences can become very long. The rewrite pattern's **`:generations`** initialization can be used to control the maximum number of generations that the pattern will produce. If a rewrite pattern reaches this optional limit it will simply cycle over the last generation from that point onward.

The rewrite pattern supports two different styles of rule specification. *Context free* rules are rules that are associated with specific elements in the pattern. Context sensitive rules are rules that are applied to a set of elements at the same time.

### Context Free Grammars

In a context free rewrite pattern each element of data is represented by a node in the pattern. The node associates a single element with a rule to replace that element in the next generation. This type of rule is termed “context free” because the rule depends only on the node itself and not its context, or position, in the surrounding nodes in the pattern. The form of a context free node is almost identical to the graph pattern:

```
({element} [:id x] [:to {id | (id ...) | pattern | false})
```

The only difference is that the value of :to can be an id, a list of ids, a pattern that produces ids, or false. Interaction 20-8 shows the rules discussed in the preceding section implemented as a context-free rewrite pattern:

Interaction 20-8. Three generations from a context-free pattern.

```
cm> (define p
      (new rewrite :of '((a :to (b a))
                        (b :to (a a b)))
            :initially '(a b)))

cm> (next p 19)
(a b b a a a b a a b b a b a b a a a b)
cm>
```

Example 20-24 shows a rewrite pattern similar to Example 20-23 that generates a melodic line. The pattern contains two tokens +2 and -2 that are treated as *increments* between the notes of a melodic line:

Example 20-24. Rewrite melody.

```
(define (stepper end rate from)
  (let ((pat (new range :initially (keynum from)
                  :stepping (new rewrite
                              :of '((+2 :to (+2 -2 +2))
                                    (-2 :to (-2 -2 +2)))))))
    (process while (< (now) end)
             output (new midi :time (now)
                        :keynum (next pat)
                        :duration rate)
             wait rate)))
```

Example 20-24 uses the same grammar that the composer Tom Johnson used his piece *Formulas for String Quartet*. Given an alphabet of +2 and -2 the pattern rewrites each +2 with the sequence +2 -2 +2 and -2 with -2 -2 +2. In our example we interpret +2 or -2 as whole step up or down. The range pattern in the example describes a "walk" that starts with a particular key number *from* and proceeds according to the steps returned by the rewrite pattern. The range pattern initialization of the offset for the walk just one time, and the **`:stepping`** property causes the range's increment to be read in parallel with the offset. We listen to four simultaneous versions of the stepper process, each with a different rhythm and pitch offset.

Interaction 20-9. Listening to four voice stepper.

```
cm> (events (list (stepper 20 (rhythm 's 100) 'c5)
                  (stepper 20 (rhythm 'te 100) 'f4)
                  (stepper 20 (rhythm 'e 100) 'bf3)
                  (stepper 20 (rhythm 'tq 100) 'ef3))
            "pat-19.midi")
"pat-19.midi"
cm>
```

- pat-19.midi Four channels.

## Context Sensitive Rules

Context sensitive rules are rules that can apply to any number of rewrite terms. Since there can be more or fewer rules than the terms they act upon, a context sensitive pattern separates the nodes, or terms, of the pattern from the rules that act upon them to produce a new generation. In a context sensitive rewrite pattern, each node in the current generation is matched against the rule set to find successor rules to apply to the node. The first rule that matches is triggered and the ids in the right-hand side of the rule provide the names of the successor nodes in the next generation.

Node specification for the context sensitive patterns is similar to the first method except that:

- the **`->`** option is not part of the node declaration.

- if the node's **`id`** value is the same as the element itself then the element itself can be specified in place of the node.

Each rule in the second style of rule specification is a list of the form:

```
(id... -> id ...)
```

The **`->`** rule operator divides each rule into two sides. The left-hand side of the rule defines a "matching target" and the right-hand side defines the rewrite succession. Either or both sides may contain more than one id. If the left-hand side of the rule is a single id then the rule matches any node with the same id. If the left-hand side has more than one id (a context-sensitive rule) then the rule matches if the *strict predecessor* in the left-hand side matches the current node and the adjacent nodes in the generation match the ids adjacent to the strict predecessor. The strict predecessor id is notated as a list. Every context rule must contain exactly one strict predecessor in its left hand side.

```
(1 (1) 2 -> 3)
(1 * (2) -> 1 2)
(5 (3) 3 4 -> )
```

The first rules specifies that node 1 rewrites to node 3 wherever 1 is preceded by itself and followed by 2 in the current generation. The second rule says that node 2 rewrites to 1 and 2 whenever 1 occurs two positions earlier in the current generation. The third rule means node 3 rewrites to nothing if preceded by 5 and followed by itself and 4 in the current generation. Note that the right-hand side may be empty and that the left-hand side may contain the wild card * which matches any single element in the current generation.

Example 20-25. Harmonies from Context sensitive rewrite.

```
(define (lsystem-harmonies len rhy dur)
  (let ((lsys (new rewrite :of '(((0 4 7) :id f)
                                 ((0 3 7) :id g)
                                 (+1 :id +)
                                 (-1 :id -))
                   :initially '(f + f - g - f)
                   :rules '((f -> f f - g - - g - f)
                            (g -> g f + f))))
        (term false)
        (knum 60))
    (process repeat len
             do
             (set! term false)
             (loop until term
                   for x = (next lsys)
                   do
                   (if (list? x)
                     (set! term x)
                     (set! knum (fit (+ knum x) 48 72))))
             each c in (transpose term knum)
             output (new midi :time (now)
                         :keynum c
                         :duration dur
                         :amplitude .7)
             wait rhy)))
```

The **`lsystem-harmonies`** process uses a context sensitive rewrite pattern to generate a harmonic grammar of of chords and offsets that are transposed to specific pitches by the process. The variable *lsys* holds the pattern and *knum* holds the current transposition level for the process. There are four nodes in the rewrite pattern, the first two define major and minor chords (0 4 7) and (0 3 7) that are named F and G in the pattern, respectively. The other two nodes define transposition increments of(+1 and -1 and named + and -, respectively. The list of terms specified to **`:initially`** sets the first generation of the pattern to a specific list of node ids. Two rules are specified to the pattern, the first rule rewrites F terms and the second G terms. The process uses a loop to read successive terms from the pattern until a major or minor chord list has been produced. If -1 or 1 is encountered during this process the loop adjusts the current offset level *knum* by that amount. The **`fit`** function is used to ensure that this value remains within a two octave range from key number 48 to 72. Once the loop has produced the next chord list from the pattern the **`each`** clauses iterates every value in the chord transposed to the current offset value in *knum* and outputs a midi object to play it.

Interaction 20-10. Listening to context-sensitive rewrite.

```
cm> (events (lsystem-harmonies 200 .2 .35)
            "pat-20.midi")
"pat-20.midi"
cm>
```

- pat-20.midi Four channels.

## Gloriette for John Cage

Gloriette for John Cage (1994) is a completely pattern based algorithmic composition. It was written in 1994 as a memorial tribute to John Cage and was commissioned for the Busy Drone mechanical organ at the Stedelijk Museum in Amsterdam. The Busy Drone organ reads a large cardboard score that functions just like a piano roll except that the size and weight of the cardboard restricts the mechanical organ to pieces that last only a few minutes in duration. Keep in mind as you listen to the piece that every note you hear was punched into the heavy cardboard score by hand!

In keeping with the Cage's interest in aleatoric music, the main algorithm in the piece uses a chance process in which the likelihood of the musical notes C,A, G, and E gradually increases as a function of time, causing the composer's name to slowly emerge out of a background of G Dorian. The rhythmic scheme and number of voices are similarly inspired by the composer's name. Gloriette for John Cage was composed at the Zentrum fuer Kunst und Medientechnologie in Karlsruhe, Germany and has been included on several CDs.

- gloriette.mp3 Gloriette for John Cage, performed by the Busy Drone mechanical organ at the Multimediale II Festival in Karlsruhe Germany, 1994.

The code for a simplified version of this composition is show in Example 20-26.

Example 20-26. Probability as a function of time.

```
(define (cage offset)
  (let* ((w 0)                       ; weight of CAGE
         (d (pval w))                ; delay eval of w
         (p (new random
              :of `((g3 :weight ,d) ; delayed weight of G
                    (a3 :weight ,d) ; delayed weight of A
                    bf3
                    (c4 :weight ,d) ; delayed weight of C
                    d4
                    (e4 :weight ,d) ; delay weight of E
                    f4
                    (g4 :weight ,d) ; delay weight of G
```

```
                   (r :max 1 :weight .25)) ; add rests
             :for 1))
        (q (new random
             :of (list 1/16 1/8
                       (new cycle :of 1/32 :for 2 )))))
    (process repeat 100
             for n = (next p)
             for r = (rhythm (next q) 65)
             for i from 0
             ;; interpolate new weight value for CAGE notes.
             set w = (interp i 0 .5 90 4)
             output (new midi :time (now)
                         :duration r
                         :keynum (transpose n offset))
             wait r)))
```

The **`cage`** process demonstrates how the probability of a **`random`** pattern can be controlled as a function of time. Over the course of 100 iterations the notes C,A,G and E gradually gain more probability than other notes in the G Dorian collection. The *`w`* variable controls the probability of the CAGE notes being selected relative to the others. The variable *`d`* is a **`pval`** object that delays the evaluation of *`w`* and allows it to be placed inside the random pattern of notes. Each iteration of the process reads a note from the pattern and then sets *`w`* to a new interpolated value. The interpolation at the start of the process sets *`w`* to .5, so the CAGE notes at the start of the process are only half as likely to be selected as the other notes in G Dorian. The weight variable *`w`* is set to a new weight value for the CAGE notes on each iteration of the process, so that each time a note is read from the pattern the value of the CAGE notes will have increased slightly over the previous iteration. By the 90th iteration the CAGE notes will be four times a likely as the others to be selected. Interaction 20-11 generates four **`cage`** processes in several different octaves, each offset by a half note a tempo 65.

Interaction 20-11. Listening to the Cage algorithm.

```
cm> (events (loop for o in '(-12 0 12 24)
                  collect (cage o))
            "pat-21.midi"
            (loop for r in '(0 h w w.)
                  collect (rhythm r 65)))
"pat-21.midi"
cm>
```

- pat-21.midi

# Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file patterns.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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06 May 2003

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# 21 *Etudes*, Op. 7: Automatic Jazz

In this Etude we examine a program that performs pattern-based jazz improvisation. The code is based on a program originally written by Eric Flister at CCRMA, Stanford University, as a project for his undergraduate computer music class. His original code has been slightly simplified and adapted to work with General MIDI instruments. His improviser generates music for a jazz combo made up of piano, acoustic bass and percussion parts. Our first step will be to learn how to specify the appropriate MIDI *program changes* to establish the Jazz Combo instruments on three channels of the synthesizer.

## MIDI Messages

Until this point we have worked exclusively with **`midi`** events — object-oriented representations of MIDI note information. It may surprise you to learn that these objects are not actually defined by the MIDI protocol! MIDI events are high-level abstractions in Common Music that represent a pair of low-level *MIDI messages*. A MIDI message is a packet of numbers sent to the synthesizer to control its operation. The two messages that a MIDI event represents are called *Note On* and *Note Off* messages, and these are what MIDI keyboards send when keys are depressed and released. The MIDI specification defines many different types of messages that collectively implement all of the ways that a MIDI synthesizer can be controlled. A MIDI message is a low-level and efficient representation of this control information. A message consists of a series of *bytes*, where each MIDI byte holds up to 7 *bits* of information, enough to represent $2^7$ different states, or the series of integers ranging from 0 to 127.

Common Music represents a MIDI message as a specially formatted integer containing up to 3 MIDI bytes of information. All of the different message types defined in the MIDI protocol are supported. Each of these messages is implemented by a constructor function, a predicate, and one or more byte accessors, if appropriate. Interaction 21-1 demonstrates the creation and display of low-level MIDI Note On and Note off messages.

Interaction 21-1. Creating and displaying MIDI messages.

```
cm> (make-note-on 0 60 90)
155226
cm> (midi-print-message 155226)
#<Note-On 0 60 90>
155226
cm> (make-note-off 9 48 127)
2496639
cm> (midi-print-message 2496639)
#<Note-Off P "Hi-Mid-Tom" 127>
2498175
cm>
```

The Note On message was created with a channel value of 0, key number 60 and key velocity 90. The utility function **`midi-print-message`** "decodes" low-level midi messages into a readable text format. The Note Off message was created with a channel value of 9, the channel that General MIDI reserves for the percussion map (in zero based counting). The **`midi-print-message`** function prints P for the channel number and also displays the sound associated with the key number when a Note On or Note Off message references the drum map.

### Program Changes

A MIDI *program change* assigns a MIDI instrument (sometimes called a program, patch or tone) to a specific channel of the synthesizer. Program change values between 0 and 127 can be sent to any non-percussion channel of a General MIDI synthesizer. The program change values needed for our Jazz Combo are 0, the Acoustic Grand

Piano, and program 32, the Acoustic Bass. A program change is not necessary to select percussion parts since General MIDI already reserves channel 9 for the drum map. Interaction 21-2 shows the creation of our MIDI program changes using the **make-program-change** constructor.

Interaction 21-2. Program changes for Jazz Instruments

```
cm> (make-program-change 0 0)
196608
cm> (midi-print-message 196608)
#<Program-Change 0 "Acoustic-Grand-Piano">
196608
cm> (make-program-change 1 32)
462848
cm> (midi-print-message 462848)
#<Program-Change 1 "Acoustic-Bass">
462848
cm>
```

The constructor **make-program-change** takes a channel number and a program change value and returns a MIDI program change message. **midi-print-message** includes the name of the instrument in its display.

## The midimsg Object

Low-level MIDI messages are just specially formatted numbers. Common Music provides the **midimsg** object to represent these messages in high-level score generation. A **midimsg** object can hold any MIDI message defined in the MIDI 1.0 specification and enables these messages to be computed along with MIDI notes in score output:

[Class]
 **midimsg**

A **midimsg** is an object-oriented representation of a low-level MIDI message. The midimsg event supports the following initializations:

:time *number*
 The score time of the midimsg.
:msg *integer*
 The 28 bit integer MIDI message.
:data *list*
 A list of any additional data bytes required by *message*.

We are now ready to define program changes for setting up the synthesizer to play the Jazz Combo. We use two **midimsg** objects, the first sends a program change 0 on channel 0 to select Acoustic Grand Piano and the second sends program change 32 on channel 1 to select an Acoustic Bass. We use a **seq** object to hold these events so that we can mix them into the output when it comes time to generate the score.

Example 21-1. A seq of program changes for the Jazz Combo

```
(new seq :name 'combo-setup
     :subobjects
     (list (new midimsg :time 0
                :msg (make-program-change 0 0))
           (new midimsg :time 0
                :msg (make-program-change 1 32))))
```

We can test the setup by writing a score containing these program changes and two test tones:

Interaction 21-3. Testing the program changes.

```
cm> (events (list #&combo-setup
                  (new midi :time 1 :channel 0
                       :keynum 60)
                  (new midi :time 3 :channel 1
                       :keynum 48))
            "pctest.midi")
"pctest-1.midi"
cm>
```

- pctest-1.midi (two channels, program changes)

Playing the file should cause the Acoustic Grand Piano to play on channel 0 and the Acoustic Bass on channel 1.

## The Automatic Jazz Program

The automatic jazz program is implemented by a "conductor" process that executes for a specified number of measures and sprouts piano, percussion and bass processes to improvise each measure. We will examine each subprocess in detail before looking at the main control algorithm. Each component processes uses data passed to it from the main process. This data includes the jazz scale to improvise with, a transposition level for the jazz scale, a tempo factor and an amplitude level. The amplitude level is adjusted on a per measure basis by the main conductor process but the other data are defined as global variables that can be adjusted and redefined by the composer.

Example 21-2. Global control data for the jazz improviser.

```
(define jazz-scale ; dorian with decorated octave
  '(0 2 3 5 7 9 10 12 14))

(define jazz-changes ; key changes
  '(bf3 ef4 bf3 bf ef4 ef bf3 bf f4 ef bf3 bf))

(define jazz-tempo 120)
```

### The Percussion Parts

The percussion parts for the Jazz Combo consist of two ride cymbals, a high hat, snare and bass drums. We will introduce these parts in their order of complexity, from simplest to most difficult. As mentioned in the previous section, each percussion sound is addressed by a specific key number in the General MIDI Drum Map. Example 21-3 contains the definition of the percussion sounds we will use:

Example 21-3. Percussion key numbers in the GM drum map.

```
(define hihat 42) ; Closed Hi Hat
(define snare 40) ; Electric Snare
(define bdrum 35) ; Acoustic Bass Drum
(define ride1 51) ; Ride Cymbal 1
(define ride2 59) ; Ride Cymbal 2
```

Consult the appendix "General MIDI Instruments" for a complete listing of the Drum Map.

### The High Hat

The High Hat percussion process is very simple, it just plays the High Hat on the second and fourth beats of every measure. Each hit sounds for the duration one triplet eighth note:

Example 21-4. The High Hat process.

```
(define (jazz-high-hat tmpo ampl)
  ;; generate a 4/4 measure of high-hat
  (let ((rhy (rhythm 'q tmpo))
        (dur (rhythm 't8 tmpo))
        (amp (amplitude 'mp))
        (pat (new cycle
               :of `(r ,hihat r ,hihat ))))
    (process repeat 4
             output
             (new midi :time (now)
                  :keynum (next pat)
                  :channel 9 :duration dur
                  :amplitude (* amp ampl))
             wait rhy)))
```

The variable *pat* holds a cyclic pattern that rests (r) on the first and third beats, and plays the key number of *hihat* on the other second and fourth beat.

We now listen to eight measures of the High Hat. Since the process computes only one measure, we sprout eight versions of the process and offset each copy by two seconds, exactly the duration of the combo's 4/4 measure at tempo 120.

Interaction 21-4. Playing eight measures of the high hat.

```
cm> (events (loop repeat 8 collect (jazz-high-hat 120 1))
            "perc.midi"
            '(0 2 4 6 8 10 12 14))
"perc-1.midi"
cm>
```

- perc-1.midi

### The Jazz Drums

The **jazz-drums** process is also quite simple. It divides each beat into a triplet quarter followed by a triplet eight and then, on each note, randomly selects between playing the snare, the bass drum or resting.

Example 21-5. Jazz drums

```
(define (jazz-drums tmpo ampl)
  (let ((rhys (new cycle :of '(t4 t8)))
        (amps (new random :of '(f (ffff :weight .1))))
        (knums (new random
                 :of `((r :weight .25) ,snare ,bdrum))))
    (process repeat 8
             for k = (next knums)
             for a = (amplitude (next amps))
             for r = (rhythm (next rhys) tmpo)
             output
```

```
              (new midi :time (now)
                   :keynum k :channel 9
                   :duration r
                   :amplitude (* a ampl))
              wait r)))
```

The *knums* variable holds the random pattern that selects between the snare, the bass drum or the rest value **`r`**. The probability of resting is .25, which means that resting is only one quarter as likely as playing either drum. The amplitudes of the drums are generally forte but one tenth of the time the process produces a very loud tone.

Interaction 21-5. Playing eight measures of the drums.

```
cm> (events (loop repeat 8
              collect
              (list (jazz-high-hat 120 .99)
                    (jazz-drums 120 .99)))
            "perc.midi"
            '(0 2 4 6 8 10 12 14))
"perc-2.midi"
cm>
```

- perc-2.midi

### The Jazz Cymbals

The **`jazz-cymbals`** process is slightly more complex than either the high hat or the drums. The cymbals process performs a constant stream of triplet eighths in which the *ride1* cymbal is played on the beginning or every quarter note. The second and third triplets of each beat are either rests or a random choice between *ride1*, *ride2* or a rest. Table 21-1 shows the beat map for a measure of the process, where 1 means the *ride1* cymbal is played, r means a rest and x means a random choice between *ride1*, *ride2* or a rest:

Table 21-1. Beat map for one measure of the ride cymbals.

| **triplet** | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| **cymbals** | 1 | r | x | 1 | r | 1 | 1 | x | x | 1 | x | 1 |

The random elements marked x in Table 21-1 are created by a helper function called **`or12r`**:

Example 21-6. Random cymbal element.

```
(define (or12r wt)
  ;; wt is weight of resting relative to playing
  ;; r1 pattern plays ride1 or rests
  ;; r2 pattern plays ride2 or rests
  (let ((r1 (new random
              :of `(,ride1 (r :weight ,wt) )
              :for 1))
        (r2 (new random
              :of `(,ride2 (r :weight ,wt) )
              :for 1)))
    ;; return random pattern that slightly
    ;; prefers r1 pattern over r2 pattern
    (new random
      :of `((,r1 :weight 1.5)
            ,r2)
      :for 1)))
```

The **`or12r`** function returns a pattern that randomly selects between two sub-patterns, *r1* or *r2*. Each sub-pattern, in turn, randomly selects between playing a cymbal (ride1 in the case of *r1* and ride2 in the case of *r2*) or a rest. The probability weight *wt* is the probability of playing a rest relative to playing either one of the ride cymbals. The pattern returned by **`or12r`** assigns the overall weight of 1.5 to the *r1* sub-pattern, which means that probability of *r1* relative to *r2* is 3:2.

Given this helper function, the **`jazz-cymbals`** process creates the pattern described in Table 21-1 to improvise one measure of music:

Example 21-7. Jazz cymbals.

```
(define (jazz-cymbals tmpo ampl)
  (let* ((rhy (rhythm 't8 tmpo))
         (amps (new cycle
                 :of '(mf mp fff f mp ffff mf
                       mp fff f mp ffff)))
         (knums (new cycle
                  :of (list ride1 'r (or12r 5)
                            ride1 'r ride1
                            ride1 (or12r 7) (or12r 7)
                            ride1 (or12r 3) ride1))))
    (process repeat 12
             for k = (next knums)
             for a = (amplitude (next amps))
             output
             (new midi :time (now)
                  :keynum k
                  :channel 9
                  :duration rhy
                  :amplitude (* a ampl ))
             wait rhy)))
```

The *knums* variable holds the random pattern that produces ride1, ride2 or the random choices created by the helper function. We can now listen eight bars of all the percussion together:

Interaction 21-6. Listening to the percussion sounds.

```
cm> (events (loop repeat 8
              collect
              (list (jazz-high-hat 120 1)
                    (jazz-drums 120 1)
                    (jazz-cymbals 120 1)))
            "perc.midi"
            '(0 2 4 6 8 10 12 14))
"perc-3.midi"
cm>
```

- perc-3.midi

## The Jazz Piano Part

The jazz piano plays "improvised" jazz chords based on a pattern of root changes and a scale pattern that is transposed to each root. The piano randomly choose between playing triplet eighths or straight eights for a given measure.

Example 21-8. The jazz piano process.

```
(define (jazz-piano scale on tmpo ampl)
  ;; generate a measure of jazz harmony.
  ;; measure contains either 8 or 12 notes.
  (let* ((reps (odds .65 8 12))
         (rhys (if (= reps 8)
                   (new cycle :of '(t4 t8))
                   (new cycle :of 't8)))
         (amps (if (= reps 8)
                   (new random
                     :of (list (new cycle :of '(mp f))
                               (new cycle :of '(mf ff))))
                   (new random
                     :of (list (new cycle :of '(mp p f))
                               (new cycle :of '(mf mp ff))))))
         (knms (new random
                   :of `((,(new heap :of scale
                                :for (new random
                                       :of '(1 2 3 4)))
                          :weight ,(new random
                                     :of '(1.15 1.65)))
                         r))))
    (process repeat reps
             for r = (rhythm (next rhys) tmpo)
             for a = (amplitude (next amps))
             for l = (transpose (next knms true) on)
             each k in l
             output (new midi :time (now)
                          :channel 0 :keynum k
                          :duration r
                          :amplitude (* a ampl))
             wait r)))
```

The **`jazz-piano`** process creates a measure of jazz harmonies given a *scale*, a transposition level *on*, a tempo factor *tmpo* and an amplitude level *ampl* that scales the local amplitude value produced in the process. Each measure of the **`jazz-piano`** part will contain *reps* number of notes as determined by the **`odds`** function, which in this process chooses 8 notes per measure approximately 65% of the time, otherwise the measure will contain 12 notes. The *rhys* variable is set to a pattern of rhythms that depends on the value of *reps*. If the piano plays 8 notes in the measure then the rhythmic pattern for the process will consists of triplet quarter (tq) followed by a triplet 8ths (te), otherwise the piano will play twelve triplet 8ths. The harmonies are generated by a random pattern that selects between a rest and a heap of notes created from the *scale* that was passed into process. Probability of choosing a note is either 1.16 or 1.65 relative to the rest and each time the heap is selected it will generate one to four notes.

The process iterates *reps* times. On each iteration it selects a new rhythm, amplitude and list of key numbers to play. The key numbers are always returned in a list because the **`next`** function is called with its optional second argument set to true, which means that an entire period of key numbers is returned each time. This list will therefore hold between one and four key numbers on each iteration and the **`each`** clause iterates every key number *k* in the list and outputs a midi event to play it. Since the key numbers are all output at the same current time the piano produces up to four simultaneous notes on every rhythm in the measure. We now listen to the piano part in isolation. We will create four measures using the jazz-scale we defined transposed to bf3 at tempo 120.

Interaction 21-7. Playing four measures of the piano part.

```
cm> (events (loop repeat 4
                  collect (jazz-piano jazz-scale 'bf3 120 1))
            "piano.midi"
            '(0 2 4 6))
"piano-1.midi"
cm>
```

- piano-1.midi

## The Acoustic Bass Part

The acoustic bass part is perhaps the most complex in terms of its implementation. The bass part plays a melodic line built out of tones from the jazz-scale's tonic seventh chord alternating with color tones outside the tonic chord. The process first divides the jazz-scale (0 2 3 5 7 9 10 11 12 14) into two sets: the tonic set, designated "t" in the code below, contains the tonic seventh pitches 0 2, 4, 6 and 7, and the color set, designated "c", which contains the decoration pitches 1, 3, 5, 7 and 9. The bass plays a series of 12 triplets per measure, on each triplet only one of the two sets is possible. On all but the first triplet a rest is also possible. A tonic note always sounds on the 1st triplet eighth of the measure. But in the next three groups of three triplets there are five alternate patterns possible, only one of which is selected at random to be played in the measure. The top line in Table 21-2 shows the triplet 8th sub-groupings within the measure, and the five possible choices for each group appear on the lines below it. Each of these lines, labeled A through E, designates the probability of selection from most to least likely. Within each triplet grouping, the letter r stands for rest, t stands for a tonic note and c stands for a color note. The 11th and 12th triplets in the measure are simply random choices between either a tonic note and a rest (t|r) or a color note and a rest (c|r)

Table 21-2. Possible sub-patterns a-e within one measure's triplet beat map.

| | **1** | **2-4** | **5-7** | **8-10** | **11** | **12** |
|---|---|---|---|---|---|---|
| **a** | t | r r c | r r t | r r c | t\|r | c\|r |
| **b** | t | r r r | r r r | r t c | t\|r | c\|r |
| **c** | t | r t c | r c t | r r r | t\|r | c\|r |
| **d** | t | c t c | t c t | c t c | t\|r | c\|r |
| **e** | t | c t r | t c r | c t r | t\|r | c\|r |

The **`acoustic-bass`** process defines two helper functions to assist in pattern creation:

Example 21-9. Helper functions for the acoustic bass process.

```
(define (getset scale ints)
  ;; return the notes in scale
  ;; at the positions in ints.
  (loop for i in ints
        collect (list-ref scale i)))

(define (rancyc data prob)
  ;; create cyclic pattern to be used
  ;; as element in random pattern with weight prob
  (list (new cycle :of data) :weight prob))
```

The **`getset`** function returns the list of notes in *scale* whose indices are specified in the list *ints*. The **`rancyc`** function creates the sub-pattern groups shown in Table 21-2. It returns a pattern node whose element is a cyclic pattern with an associated probability weight *prob* in the surrounding random pattern.

The **`jazz-bass`** process uses the helper functions to create the pattern data. It first uses **`getset`** to partition the jazz-scale into two random patterns, *tonics* and *colors*. The *bmap* variable holds the "beat map" for the measure. The pattern will returns twelve values, each value is either the symbol t c or r. The t value means the process should read a note from the *tonic* pattern, c means to read from the color pattern and r means to rest. The beatmap divides the twelve triplets of the measure into pattern subgroups containing 4+3+3+1+1 values. Each of first three subgroups chooses one of five possible patterns created by the **`rancyc`** helper function.

Example 21-10. The acoustic bass process.

```
(define (jazz-bass scale on tmpo ampl)
  (let ((rhy (rhythm 'te tmpo))
        (tonics (new random
                  :of (getset scale '(0 2 4 6 7))))
        (colors (new random
                  :of (getset scale '(1 3 5 6 8))))
        (amps (new cycle
                :of '(mp p ffff fff p fff
                      mp p ffff fff mp fff)))
        (durs (new cycle :of '(tq t8 t8)))
        ;; beat map
        (bmap
         (new cycle
           :of
           (list
            ;; 5 possible patterns for 1-4
            (new random :for 1
                 :of (list (rancyc '(t r r c) 1.0)
                           (rancyc '(t r r r) .25)
                           (rancyc '(t r t c) .22)
                           (rancyc '(t c t c) .065)
                           (rancyc '(t c t r) .014)))
            ;; 5 possible patterns for 5-7
            (new random :for 1
                 :of (list (rancyc '(r r t) 1.0)
                           (rancyc '(r r r) .25)
                           (rancyc '(r c t) .22)
                           (rancyc '(t c t) .038)
                           (rancyc '(t c r) .007)))
            ;; 5 possible patterns for 8-10
            (new random :for 1
                 :of (list (rancyc '(r r c) 1.0)
                           (rancyc '(r t c) .415)
                           (rancyc '(r r r) .25)
                           (rancyc '(c t c) .11)
                           (rancyc '(c t r) .018)))
            ;; two possible values for 11
            (new random :for 1
                 :of '((r :weight 1)
                       (t :weight .25)))
            ;; two possible values for 12
            (new random :for 1
                 :of '((r :weight 1)
                       (c :weight .25)))))))
    (process repeat 12
             for x = (next bmap)
             for k = (if (equal? x 't)
                         (next tonics)
                         (if (equal? x 'c)
                           (next colors)
                           x))
             for d = (rhythm (next durs) tmpo)
             for a = (amplitude (next amps))
             output
             (new midi :time (now)      ;
                  :keynum (transpose k on)
                  :channel 1
                  :duration d
                  :amplitude (* a ampl))
             wait rhy)))
```

The process plays twelve notes per measure, on each note a value *x* is read from the beat map. If *x* is either **`t`** or **`c`** then *k* is set to a value from the *tonic* or *color* pattern, otherwise *x* is a rest (**`r`**). The key number of each **`midi`** event is determined by transposing the scale degree *k* to the transposition level *on* specified to the process. We now listen to eight measures of **`jazz-bass`** in isolation:

Interaction 21-8. Listening to eight bars of the acoustic bass.

```
cm> (events (loop repeat 8
                  collect (jazz-bass jazz-scale 'bf2 120 1))
            "bass.midi"
            '(0 2 4 6 8 10 12 14))
"bass-1.midi"
cm>
```

- bass-1.midi

## The Main Process

The main process of the automatic jazz program is quite simple: **`jazz-combo`** iterates *measures* number of times and sprouts the combo's instrument at the start of each measure. Every 12 measures **`Jazz-combo`** adjusts its overall amplitude level so that the combo's amplitude changes dynamically during the performance. Similar adjustments could be easily made to the tempo, scale and root movements of the jazz improvisation. These improvements are left to the reader to implement.

Example 21-11. The main process.

```
(define (jazz-combo measures changes tempo scale)
  (let ((roots (new cycle :of changes))
        (ampl 1))
    (process for meas below measures
             for root = (next roots)
             if (= 0 (mod meas 12))
             set ampl = (between .5 1)
             sprout (jazz-piano scale root tempo ampl)
             sprout (jazz-cymbals tempo ampl)
             sprout (jazz-high-hat tempo ampl)
             sprout (jazz-drums tempo ampl)
             sprout (jazz-bass scale (transpose root -12)
                               tempo ampl)
             wait (rhythm 'w tempo))))
```

We now generate a 24 bar improvisation from the jazz combo. Sound examples from two different runs are provided.

Interaction 21-9. Two different versions of the Jazz Combo.

```
cm> (events (list #&combo-setup
                  (jazz-combo 48 jazz-changes
                              jazz-tempo jazz-scale))
            "jazz.midi")
"jazz-1.midi"
cm> (events (list #&combo-setup
                  (jazz-combo 48 jazz-changes
                              jazz-tempo jazz-scale))
            "jazz.midi")
"jazz-2.midi"
cm>
```

- jazz-1.midi
- jazz-2.midi

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file jazz.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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# 22 *Etudes*, Op. 8: An Algorithmic Model of György Ligeti's *Désordre*

In this Etude we look at a pattern-based description of Georgi Ligeti's first Piano Etude, *Désordre*. The program examples and most of the commentary were written by Tobias Kunze at CCRMA, Stanford University. *Désordre* has been analyzed in great depth by Kinzler in György Ligeti: decision and automatism in `Désordre', 1er Étude, Premier Livre (Interface, Swets & Zeitlinger, 1991). Jane Piper Clendinning's The Pattern Meccanico Compositions of György Ligeti (Perspectives of New Music) discusses Ligeti's use of patterns in a number of different compositions.

The intent of this Etude is to consider Ligeti's composition from the perspective of algorithmic control — to arrive at some idea of what specific "algorithmic knobs" could be implemented that might offer a possible analog to the composer's actual decisions. We first present a general analysis of the Etude's essential features, and then model the composition in software using process descriptions. The model makes an honest attempt to be realistic but trades off accuracy against flexibility. After all, there is no point in implementing an algorithm that cannot be manipulated or adjusted so that it can explore different paths through a composition's theoretical parameter space.

A note regarding references to the score: since the hands are essentially independent, each point of interest in the score will be indicated by page, system and measure numbers. Page numbers are prefixed by *p*, systems by *s* and measures by *m*. For example "p3s2m1" stands for page 3, system 2. measure 1. Where necessary, an indication of the part will also be provided. In systems starting with a partial measure, the first complete measure is counted as measure 1. All numbers follow the facsimile edition of Études pour piano: premier livre. published by B. Schott's Söhne, Mainz, 1985.

## Structure

The textural make-up of the Etude is rather straightforward. Each hand is assigned its own part, both parts share a common, layered substructure and a continuously pulsating eighth-note motion. Both hands are restricted to a particular set of notes: the right hand plays only the white keys and the left hand plays only black keys. In addition to establishing an essential dramatic opposition in the piece, the partitioning also allows the relatively difficult parts to occupy the same register at certain points in the composition. In keeping with the harmonic partitioning, each part follows its own internal logic and is essentially independent from the other horizontally as well. That is, both hands are largely unsynchronized on the metrical level (although stretches with semi-synchronized phrase structures do exist.) This metrical independence is central to the composition of *DéSordre* and most of the form-defining features of the work stem from this basic independence.

The internal structure of each part is a combination of constantly rising eighth notes in the background and a rhythmically pronounced, slower foreground, all within the constant running motion of the eight notes. The foreground layer is *forte* throughout the piece and heavily accented. The foreground layer of both parts features a phrase structure that is reminiscent of traditional A-A'-B forms, where two closely related shorter phrases are followed by a longer and developmental third phrase. In addition, the phrasing hints, in spite of the distortion by the asymmetrical meter employed in each part, at an underlying simple and song-like two-beat meter. The similarly asymmetric "meter" that is induced in the backgrounds by eighth-note groups of differing lengths is relatively negligible compared to the prominent metric structure in the foreground layers.

On a formal level, the piece is structured by both the temporal behavior and pitch behavior of its two parts. A first section, 404 eighth notes long, (from the beginning of the piece to page three/system 4/measure 4) is quite static in tempo except for the slight accelerando towards the end. The next section, which extends until page 4/system 4/measure 7 (231 eighth notes in duration) accelerates the foreground parts until they are reduced to eighth notes, at which point a typical “ligetiesque” cut interrupts the lower part, which had by then hit the bottom of the piano range and resumes in the descant as well as in original foreground tempo. This third section extends to the end and is characterized by a static tempo in the upper part and a slowing down of the lower part. It measures 429 eighth notes.

## The Patterns

The foreground in both the upper and lower parts consist of cyclic repetitions of a constant, stepwise pattern. With each cycle the pattern is transposed diatonically by a constant amount. The pattern for the upper part consists of 26 steps, 7 for each of the A phrases and 12 for phrase B:

Table 22-1. Step Pattern for white keys.

| | |
|---|---|
| **Phrase A** | 0 0 1 0 2 1 -1 |
| **Phrase A'** | -1 -1 2 1 3 2 -2 |
| **Phrase B** | 2 2 4 3 5 4 -1 0 3 2 6 5 |

The pattern for the lower part has 33 steps, again 7 for each of the A phrases and 19 in the extended phrase B:

Table 22-2. Step Pattern for black keys

| | |
|---|---|
| **Phrase A** | 0 0 1 0 2 2 0 |
| **Phrase A'** | 1 1 2 1 -2 -2 -1 |
| **Phrase B** | 1 1 2 2 0 -1 -4 -3 0 -1 3 2 1 -1 0 -3 -2 -3 -5 |

There are a total of 14 cycles in the upper part, each of which is transposed *diatonically* one step upwards. Due to its greater length, there only 11 cycles in the lower part. The smaller number of cycles — which translates into fewer transpositions — is more than compensated for by (1) a two-step diatonic transposition downwards and (2) the lower part's pentatonic mode having fewer steps per octave than the heptatonic mode of the upper part. The cycles align with the score as shown in Table 22-2

Table 22-3. Cycles and transpositions for both parts.

| | **Lower Part** | | | | **Upper Part** | |
|---|---|---|---|---|---|---|
| **#** | **Transp.** | **Score** | | **#** | **Transp.** | **Score** |
| | | | **Section I** | | | |
| 1: | ds3 | p2s1m0 | | 1: | b4 | p2s1m0 |
| 2: | as2 | p2s3m5 | | 2: | c5 | p2s2m7 |
| 3: | fs2 | p3s2m2 | | 3: | d5 | p2s4m7 |
| | | | | 4: | e5 | p3s2m6 |
| | | | **Section II** | | | |
| 4: | cs2 | p3s4m4 | | 5: | f5 | p3s4m4 |
| 5: | gs1 | p4s1m3 | | 6: | g5 | p3s4m11 |
| 6: | ds1 | p4s2m3 | | 7: | a5 | p4s1m7 |
| 7: | as0 | p4s3m2 | | 8: | b5 | p4s2m5 |
| 8: | fs0 | p4s4m1[1] | | 9: | c6 | p4s3m3 |
| | | | | 10: | d6 | p4s3m10 |
| | | | **Section III** | | | |

| | | | | | |
|---|---|---|---|---|---|
| | | (p4s4m7) | 11: | e6 | p4s4m7 |
| 9: | cs5 | p5s1m2 | 12: | f6 | p5s2m4 |
| 10: | gs4 | p5s3m7 | 13: | g6 | p5s4m4 |
| 11: | ds4 | p6s2m4[2] | 14: | a6 | p6s2m4 [3] |

1. Cycle 8 in the lower part reaches the bottom of the piano range and discontinues after m. 4 in phrase B only to resume immediately in the treble clef (p4s4m7). The first note of this continuation should read fs but for obvious reasons is given as fs5 to the right hand.
2. Cycle is cut off after m. 3 in Phrase B.
3. Cycle is cut off after m. 2 in Phrase A'.

The rhythmic picture is less clear due to rasterization effects in the acceleration and deceleration of the patterns. The sequence of rhythms in the upper and lower parts read as follows:

Table 22-4. Upper part, foreground rhythms.

| **Sect. I: Little Disorder (404 eighths)** | |
|---|---|
| 3 5 3 5 5 3 7 | cycle 1 |
| 3 5 3 5 5 3 7 | |
| 3 5 3 5 5 3 3 4 5 3 3 5 | |
| 3 5 3 4 5 3 8 | cycle 2 |
| 3 5 3 4 5 3 8 | |
| 3 5 3 4 5 3 3 5 5 3 3 4 | |
| 3 5 3 5 5 3 7 | cycle 3 |
| 3 5 3 5 5 3 7 | |
| 3 5 3 5 5 3 3 4 5 3 3 5 | |
| 3 5 3 4 5 2 7 | cycle 4 (accel.) |
| 2 4 2 4 4 2 5 | |
| 2 3 2 3 3 1 1 3 3 1 1 3 | |
| **Sect. II: Rapid accelerando (231 eighths)** | |
| 1 2 1 2 2 1 3 | cycle 5 |
| 1 2 1 2 2 1 3 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 | |
| 1 2 1 2 2 1 3 | cycle 6 |
| 1 2 1 2 2 1 3 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 | |
| 1 2 1 2 2 1 3 | cycle 7 |
| 1 2 1 2 2 1 2 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 | |
| 1 2 1 2 2 1 2 | cycle 8 |
| 1 2 1 2 2 1 2 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 | |
| 1 2 1 2 2 1 2 | cycle 9 |
| 1 2 1 2 1 1 2 | |
| 1 2 1 2 2 1 1 1 2 1 1 1 | |
| | cycle 10 |

| | |
|---|---|
| 1 2 1 1 1 1 2 | |
| 1 1 1 1 1 1 2 | |
| 1 1 1 1 1 1 1 1 1 1 1 1 | |

**Sect. III: Static (429 eighths)**

| | |
|---|---|
| 3 5 3 5 5 3 8 | cycle 11 |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 3 5 5 3 3 5 | |
| 3 5 3 5 5 3 8 | cycle 12 (same) |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 3 5 5 3 3 5 | |
| 3 5 3 5 5 3 8 | cycle 13 (same) |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 3 5 5 3 3 5 | |
| 3 5 3 5 5 3 8 | cycle 14 |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 8 | (cuts off here) |

Table 22-5. Lower part, foreground rhythms.

**Sect. I: Static (404 eighths)**

| | |
|---|---|
| 3 5 3 5 5 3 8 | cycle 1 |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 3 5 5 3 3 5 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 8 | cycle 2 |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 3 5 5 3 3 5 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 8 | cycle 3 |
| 3 5 3 5 5 2 7 | (accel.) |
| 3 4 3 4 4 2 2 4 4 2 2 3 2 3 1 3 3 1 4 | |

**Sect. II: Rapid Accelerando (231 eighths)**

| | |
|---|---|
| 1 3 1 2 2 1 3 | cycle 4 |
| 1 2 1 2 2 1 3 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 1 2 1 2 2 1 3 | |
| 1 3 1 2 2 1 3 | cycle 5 |
| 1 2 1 2 2 1 3 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 1 2 1 2 2 1 2 | |
| 1 2 1 2 2 1 2 | cycle 6 |
| 1 2 1 2 2 1 2 | |
| 1 2 1 2 2 1 1 2 2 1 1 2 1 2 1 2 2 1 2 | |
| 1 2 1 2 2 1 2 | cycle 7 |
| 1 2 1 2 2 1 2 | |
| 1 2 1 2 2 1 1 2 1 1 1 2 1 1 1 1 1 1 2 | |
| 1 1 1 1 1 1 2 | cycle 8 |
| 1 1 1 1 1 1 2 | |
| 1 1 1 1 1 1 1 1 | |

**Sect. III: Allargando (429 eighths)**

| | |
|---|---|
| 5 3 3 5 3 5 3 5 5 3 8 | |

| | |
|---|---|
| 3 5 3 5 5 3 8 | cycle 9 |
| 3 5 3 5 5 3 8 | |
| 3 5 3 5 5 3 3 5 6 3 3 5 3 5 3 6 5 3 8 | |
| 3 6 3 5 5 3 9 | cycle 10 |
| 3 5 3 5 6 3 8 | |
| 3 5 3 6 5 3 3 5 6 3 3 5 3 5 3 6 5 3 9 | |
| 3 7 3 8 9 3 13 | cycle 11 |
| 3 11 3 21 | |

## The Model

The brief analysis presented in the previous section will allow us to translate the combined foreground structure of *Désordre* into a CM pattern description. We begin by defining an 8th note pulse for the piece according to the specification in the score. A helper function is also defined that will allow us to convert from duration values expressed as number of 8th notes into time in seconds:

Example 22-1. Global tempo and amplitude.

```
(define eighth-pulse (rhythm 'e 76.0 'w))

(define (eighth-time number-of-eighths)
  ;; Convert duration in eighth notes
  ;; to time in seconds
  (* eighth-pulse number-of-eighths))

(define fg-amp .7)  ; foreground amp

(define bg-amp .5)  ; background amp
```

The global variable *eighth-pulse* holds the duration in seconds of an 8th note pulse at tempo 76 with a whole note beat. This value works out to 15/152, or approximately 0.098 seconds per 8th note. The function **eighth-time** translates a specified *number-of-eighths* into time in seconds by multiplying that number by the speed of one eighth note. The variables *fg-amp* and *bg-amp* define global amplitude levels for the foreground and background processes, respectively. These can be adjusted as needed to increase or decrease the amplitude level of the foreground relative to the background.

## Upper Foreground Materials

As suggested by our analysis in the preceding section, the algorithms to implement the upper and lower parts to *DéSordre* will be driven by four sets of data: the part's mode, its step pattern, its transposition pattern, and its rhythmic pattern.

### Upper Foreground Melody

We examine the melodic materials first. Example 22-2 contains the definitions of the white key mode and the step pattern for the upper part. A *step* is an interval that describes *relative* melodic motion between notes, i.e. the intervallic distances between successive notes in a sequence.

Example 22-2. Upper foreground mode and step patterns.

```
(define white-mode
  (new mode :steps '(c d e f g a b c)))
```

```
(define white-fg-steps
  '(0  0  1  0  2  1 -1            ; Phrase a
    -1 -1  2  1  3  2 -2           ; Phrase a'
    2  2  4  3  5  4 -1 0 3 2 6 5)) ; Phrase b

(define (make-white-fg-notes note)
  ;; convert note in MIDI scale to modal equivalent
  (let ((step (keynum note :to white-mode)))
    ;; pattern of steps is offset
    ;; from a constantly rising offset
    (new transposer
      :of (new cycle :of white-fg-steps)
      :by (new range :initially step :by 1))))
```

The *white-mode* global variable is set to a mode object that implements the white key mode for the upper part. Each octave has only eight tones, and adjacent key numbers in the mode will be either one or two semi-tones apart in the chromatic scale. The *white-fg-steps* variable contains the sequence of *step motion* that describes the upper part. Each step in *white-fg-steps* describes the *relative* distance from the previous step. For example, if the initial position for the step pattern were c4 then the steps (0 0 1 0 2) would produce the notes (c4 c4 cs cs ds) in the chromatic scale but (c4 c4 d4 d4 e4) in *white-mode*.

The **`make-white-fg-note`** function returns a pattern that implements the melodic foreground of the upper part. The starting point for the melody is the *note* specified to the function which is converted to its key number equivalent in *white-mode*. The **`transposer`** pattern transposes a cycle pattern of foreground steps (intervals) onto the diatonic degrees of the white mode.

## Upper Foreground Rhythmic Pattern

For reasons of clarity the rhythmic pattern for the upper part is presented here in its entirety even though is is fairly long. It would certainly be possible to define this pattern in terms of repeated subpatterns.

Example 22-3. The upper foreground rhythmic pattern.

```
(define (make-white-fg-rhythms)
  (new cycle
    :of `(3 5 3 5 5 3 7             ; cycle 1
          3 5 3 5 5 3 7
          3 5 3 5 5 3 3 4 5 3 3 5
          3 5 3 4 5 3 8             ; cycle 2
          3 5 3 4 5 3 8
          3 5 3 4 5 3 3 5 5 3 3 4
          3 5 3 5 5 3 7             ; cycle 3
          3 5 3 5 5 3 7
          3 5 3 5 5 3 3 4 5 3 3 5
          3 5 3 4 5 2 7             ; cycle 4
          2 4 2 4 4 2 5
          2 3 2 3 3 1 1 3 3 1 1 3
          1 2 1 2 2 1 3             ; cycle 5
          1 2 1 2 2 1 3
          1 2 1 2 2 1 1 2 2 1 1 2
          1 2 1 2 2 1 3             ; cycle 6
          1 2 1 2 2 1 3
          1 2 1 2 2 1 1 2 2 1 1 2
          1 2 1 2 2 1 3             ; cycle 7
          1 2 1 2 2 1 2
          1 2 1 2 2 1 1 2 2 1 1 2
          1 2 1 2 2 1 2             ; cycle 8
          1 2 1 2 2 1 2
          1 2 1 2 2 1 1 2 2 1 1 2
          1 2 1 2 2 1 2            ; cycle 9
          1 2 1 2 1 1 2
          1 2 1 2 2 1 1 1 2 1 1 1
          1 2 1 1 1 1 2            ; cycle 10
          1 1 1 1 1 1 2
          1 1 1 1 1 1 1 1 1 1 1 1
          ,(new cycle
             :of
             (new cycle
               :of
               '(3 5 3 5 5 3 8   ; cycle 11-14
                 3 5 3 5 5 3 8
                 3 5 3 5 5 3 3 5 5 3 3 5))
             :for 3)
          3 5 3 5 5 3 8            ; cycle 14
          3 5 3 5 5 3 8
          3 5 3 5 5 3)))           ; cuts off here
```

## Lower Foreground Materials

The algorithm for the lower foreground part is driven by the same types of data. However, since it is eventually shifter into the treble clef after it hits the bottom of the keyboard (p4s4m7), its transposition pattern adds a “warp” that transposes the whole pattern up five octaves, or 20 black mode steps (see the *warp-point* variable in Example 22-4).

### Lower Foreground Melody

Our lower foreground melody is defined in a manner analogous to the upper foreground, except of course that the pattern and mode are quite different. The implementation is made slightly more complex because the lower pattern “warps”, or shifts, when it reaches the bottom of the keyboard by immediately jumping back up to the middle register of the keyboard.

Example 22-4. Lower foreground mode and step patterns.

```
(define black-mode
  (new mode :steps '(cs ds fs gs as cs)))

(define black-fg-steps
  '(0  0  1  0  2  2  0               ; Phrase a
    1  1  2  1 -2 -2 -1               ; Phrase a'
    1  1  2  2  0 -1 -4 -3  0         ; Phrase b
    0 -1  3  2  1 -1  0 -3 -2 -3 -5))

(define (make-black-fg-notes note)
  (let* ((cycle-length (length black-fg-steps))
         (warp-point (+ (* cycle-length 7)
                        7 7 8))
         (step (keynum note :to black-mode)))
    (new transposer
      :of
      (new transposer
        :of (new cycle
              :of black-fg-steps)
        :stepping (new line
                    :of
                    (list (new cycle :of 0
                                :for warp-point)
```

```
                           20)))
      :by (new range :initially step :by -2))))
```

The *warp-point* variable specifies the point at which the lower process jumps from the bottom of the keyboard up to the middle register. This is implemented using two **`transposer`** patterns. The inner transposer replicates each cycle of the step relative to an offset generated from a **`line`** pattern. A line produces elements in sequential order but sticks on the last value. Thus the line specified in Example 22-4 generates no offset (returns 0) until *warp-point* but from that point onward produces the warp transposition 20, thus transposing each value produced from *black-fg-steps* cycle by 20 from that point onward. The surrounding pattern then applies the descending stepwise transposition (-2) to each period of the inner pattern.

### Lower Foreground Rhythm

The definition of the lower foreground rhythmic pattern is derived directly from the data in Table 22-4. There is enough inner repetition to make the definition of subpatterns worthwhile.

Example 22-5. The lower foreground rhythmic pattern.

```
(define (make-black-fg-rhythms)
  (new cycle
    :of '(3 5 3 5 5 3 8    ; cycle 1
          3 5 3 5 5 3 8
          3 5 3 5 5 3 3 5  5 3 3 5 3 5 3 5 5 3 8
          3 5 3 5 5 3 8    ; cycle 2
          3 5 3 5 5 3 8
          3 5 3 5 5 3 3 5  5 3 3 5 3 5 3 5 5 3 8
          3 5 3 5 5 3 8    ; cycle 3
          3 5 3 5 5 2 7
          3 4 3 4 4 2 2 4  4 2 2 3 2 3 1 3 3 1 4
          1 3 1 2 2 1 3    ; cycle 4
          1 2 1 2 2 1 3
          1 2 1 2 2 1 1 2  2 1 1 2 1 2 1 2 2 1 3
          1 3 1 2 2 1 3    ; cycle 5
          1 2 1 2 2 1 3
          1 2 1 2 2 1 1 2  2 1 1 2 1 2 1 2 2 1 2
          1 2 1 2 2 1 2    ; cycle 6
          1 2 1 2 2 1 2
          1 2 1 2 2 1 1 2  2 1 1 2 1 2 1 2 2 1 2
          1 2 1 2 2 1 2    ; cycle 7
          1 2 1 2 2 1 2
          1 2 1 2 2 1 1 2  1 1 1 2 1 1 1 1 1 1 2
          1 1 1 1 1 1 2    ; cycle 8
          1 1 1 1 1 1 2
          1 1 1 1 1 1 1 1  5 3 3 5 3 5 3 5 5 3 8
          3 5 3 5 5 3 8    ; cycle 9
          3 5 3 5 5 3 8
          3 5 3 5 5 3 3 5  6 3 3 5 3 5 3 6 5 3 8
          3 6 3 5 5 3 9    ; cycle 10
          3 5 3 5 6 3 8
          3 5 3 6 5 3 3 5  6 3 3 5 3 5 3 6 5 3 9
          3 7 3 8 9 3 13   ; cycle 11
          3 11 3 21)))     ; cuts off here
```

## Foreground Process Description

We are now ready to define a process that will play our foreground material:

Example 22-6. Foreground process.

```
(define (fg-mono mode keynums rhythms)
  (process for key = (keynum (next keynums)
                             :from mode)
           for rhy = (eighth-time (next rhythms))
           output (new midi :time (now)
                       :keynum key
                       :amplitude fg-amp
                       :duration rhy)
           wait rhy
           until (eop? rhythms)))
```

The **`fg-mono`** process takes a *mode*, a pattern of *keynums* and a pattern of *rhythms*. The process generates a melody with every keynum *key* and rhythm *rhy* in the patterns for one rhythmic period. Since the pattern *keynums* that is passed to the function describes relative intervallic motion within a particular mode, the **`:from`** keyword argument to **`keynum`** is used to convert scaler positions in *mode* into their equivalent positions in the standard chromatic scale used by MIDI. Since this mode is passed into the process, **`fg-mono`** is a completely generic description that can be used to generate both the upper and lower foreground parts. We now listen to each foreground part, first individually and then mixed together.

Interaction 22-1. Play upper foreground.

```
cm> (events (fg-mono white-mode
                     (make-white-fg-notes 'b4)
                     (make-white-fg-rhythms))
            "ligeti.midi")
"ligeti-1.midi"
cm>
```

- ligeti-1.midi

Interaction 22-2. Play lower foreground.

```
cm> (events (fg-mono black-mode
                     (make-black-fg-notes 'ds4)
                     (make-black-fg-rhythms))
            "ligeti.midi")
"ligeti-2.midi"
cm>
```

- ligeti-2.midi

Interaction 22-3. Play both foregrounds together.

```
cm> (events (list
             (fg-mono white-mode
                      (make-white-fg-notes 'b4)
                      (make-white-fg-rhythms))
             (fg-mono black-mode
                      (make-black-fg-notes 'ds4)
                      (make-black-fg-rhythms)))
            "ligeti.midi")
"ligeti-3.midi"
cm>
```

- ligeti-3.midi

## Enhancement 1

The first improvement to our model will add octaves to the foreground and simulate a background layer by filling in the 8th note pulse using randomly selected mode notes centered around each of the foreground notes.

Example 22-7. Add octaves and a fake background

```
(define (desordre-w/octaves mode ntes rhys)
  (let ((fg-time 0)
        (fg-eighths 0)
        (mode-deg 0))
    (list
     ;; foreground process
     (process with dur and key
              set fg-time = (now)
              set mode-deg = (next ntes)
              set key = (keynum mode-deg :from mode)
              set fg-eighths = (next rhys)
              set dur = (eighth-time fg-eighths)
              output (new midi
                       :time fg-time
                       :keynum key
                       :duration (- dur .01)
                       :amplitude fg-amp)
              output (new midi
                       :time fg-time
                       :keynum (+ key 12)
                       :duration (- dur .01)
                       :amplitude fg-amp)
              wait dur
              until (eop? rhys))
     ;; background process fills in 8th note pulses with
     ;; randomly selected mode notes based on current
     ;; foreground note.
     (let ((pat (new range
                  :from (pval mode-deg)
                  :stepping (new random
                              :of
                              '((1 :weight 3)
                                2
                                (3 :weight .5
                                 :max 1)))
                  :for (pval fg-eighths))))
       (process repeat 1064
                for k = (keynum (next pat)
                                :from mode)
                unless (or (= (now) fg-time)
                           (not (<= 0 k 127)))
                output (new midi
                         :time (now)
                         :keynum k
                         :duration (- eighth-pulse
                                      .01)
                         :amplitude bg-amp)
                wait eighth-pulse)))))
```

The **`desordre-w/octaves`** process takes the same arguments as our original **`play-fg`** process but its definition is more involved. Notice that **`desordre-w/octaves`** returns a *list* of two processes, the first produces our foreground materials and the second generates the background. The first process is equivalent to our original **`play-fg`** definition except that it outputs *two* MIDI objects per interaction to create the octave passage work. Our first process description sets two shared variables that the second process will use too. The *mode-deg* variable holds the current mode degree of the foreground and the variable *fg-eighths* is set to the duration (in eighth notes) of the note the process is currently playing.

The second process description creates a harmonic background to support the foreground material. It creates a pattern *pat* that generates random steps from whatever *mode-deg* the first process is currently playing. The **`range`** pattern produces each step using a random pattern that generates three different step sizes: steps to adjacent degrees (1), steps that skip one degree (2) and steps that skip two degrees (3). The actual intervallic distances that theses steps produce will depend on the mode to which it is applied. For example a single step (1) can produce either half steps or whole steps in our white mode but whole steps or minor thirds in our pentatonic black mode. Since the range's initial position as well as its period length depend on whatever the first process is currently generating, the pattern must uses **`pval`** to delay the evaluation of *fg-eighths* and *mode-deg* until the pattern is actually running. Otherwise, these variables would be evaluated when the pattern is *created*, not rather then when the pattern was used.

The rest of the process description is relatively straightforward. Because the foreground material extends to the top and bottom of the keyboard, our random background process could potentially generate random key number above or below the legal range of key numbers specified by MIDI. The **`unless`** clause checks to see if the current key number of the background is within the legal limits and, if not, omits that event from the score. We now listen to the octaves and background material:

Interaction 22-4. Listening to enhancement 1.

```
cm> (events (append
             (desordre-w/octaves white-mode
                                 (make-white-fg-notes 'b3)
                                 (make-white-fg-rhythms))
             (desordre-w/octaves black-mode
                                 (make-black-fg-notes 'ds3)
                                 (make-black-fg-rhythms)))
            "ligeti.midi")
"ligeti-4.midi"
cm>
```

- ligeti-4.midi

## Enhancement 2

As our final enhancement we implement processes that add more realism by simulating the foreground chords that occur in section three:

Example 22-8. Process with foreground chords.

```
(define (desordre-voices mode ntes rhys chord
                         voimap fgchan bgchan)
  (let ((moct (scale-divisions mode))
        (fg-time 0)
        (fg-8ths 0)
        (mode-deg 0))
    (list
     ;; foreground process
     (process with rhy and voi and key
              and voices =
              (new transposer
                :of (new heap :of chord
                         :for (pval voi))
                :on (pval mode-deg))
              for count from 0
              set fg-time = (now)
              set mode-deg = (next ntes)
              set key = (keynum mode-deg
                                :from mode)
              set fg-8ths = (next rhys)
              set rhy = (eighth-time fg-8ths)
              set voi = (lookup count voimap)
              output
              (new midi :time (now)
                   :channel fgchan
                   :keynum key
                   :duration rhy
                   :amplitude fg-amp)
              wait rhy
              if (equal? voi true)
              ;; add lower octave ...
              output
              (new midi :time (now)
                   :channel fgchan
                   :keynum (- key 12)
                   :duration rhy
                   :amplitude fg-amp)
              else
              ;; ... else add cluster
              each k in (next voices true)
              output
              (new midi
                :channel fgchan
                :time (now)
                :keynum (keynum k :from mode)
                :duration (- rhy .01))
              until (eop? rhys))
     ;; background process.
     (let ((notes
            (new range
              :from
              (pval (- mode-deg
                       moct))
              :stepping
              (new random
                :of '((1 :weight 3)
                      2
                      (3 :weight .5 :max 1)))
              :for (pval fg-8ths))))
       (process for i below 1064
                for k = (keynum (next notes)
                                :from mode)
                unless (or (= (now) fg-time)
                           (not (<= 0 k 127)))
                output (new midi
                         :channel bgchan
                         :time (now)
                         :keynum k
                         :duration (- eighth-pulse
                                      .01)
                         :amplitude bg-amp)
                wait eighth-pulse)))))
```

As in the first enhancement (**`desordre-w/octaves`**) our second enhancement creates a foreground and background process. The background process is essentially unchanged from the first enhancement and will not be discussed here. However, the first process is considerably more complex in our new definition because the foreground will now generate both octaves and clusters, or random chords, of notes. The intervallic content of these random clusters is specified to the process by the *chord* parameter. The value of *chords* is a list of intervals that the *voices* random pattern will transpose to the foreground's current mode degree. The density, or number of notes, in each random clusters is controlled by a “voice map” specified in the *voimap* parameter. The voice map is a list that contains *pairs* of values. The first value in each pair specifies a beat number in the score (calculated in eighth notes). The second value in each pair specifies the number notes that each cluster should contain from that beat forward until the next pair in the beat map. Since our foreground process will produce both octaves and chords, a boolean true value in the voice map will be interpreted as meaning that octaves instead of random clusters should be output. For example the beat map:

```
        (0 3 10 #t 15 6)
```

would specify 3-voice clusters starting at beat 0, octaves starting at beat 10 and 6-voice clusters staring at beat 6. The Common Music function **`lookup`** is used to look up the current voice number *voi* in *voimap*. **`lookup`** is similar to **`interpl`** in that it processes lists of pairs whose "x" value is monotonically increasing. However **`lookup`** does not perform linear interpolation, it simply returns whatever "y" value is in the list. Since no interpolation is performed the "y" value does not even have to be a numeric value.

We now listen to our final version. Foreground and background processes for both parts are given their own MIDI channels so that each individual processes can heard in isolation or in combination with other specific voices.

Interaction 22-5. Listening to enhancement 2.

```
cm> (events (append
             (desordre-voices white-mode
                              (make-white-fg-notes 'b4)
                              (make-white-fg-rhythms)
                              '(-1 -2 -3 -4 -5 -6)
                              '(0 #t 261 1 284 2 297 3)
                              0 1)
             (desordre-voices black-mode
                              (make-black-fg-notes 'ds4)
                              (make-black-fg-rhythms)
                              '(-1 -2 -3 -4)
                              '(0 #t 254 1 286 2)
                              2 3))
            "ligeti.midi")
"ligeti-5.midi"
cm>
```

- ligeti-5.midi (four channels)

# Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file ligeti.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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06 May 2003

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# 23 Spectral Composition

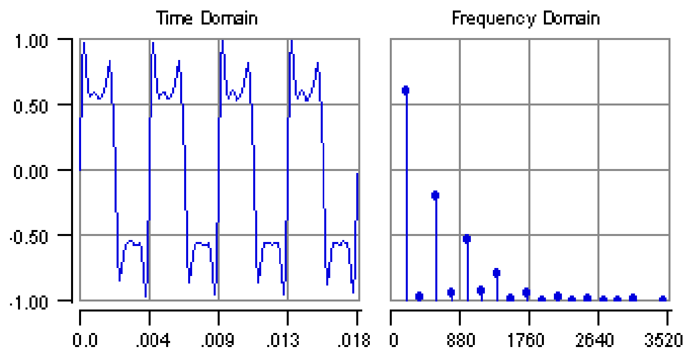
In *spectral composition* the acoustic properties of sound constitute source material for music composition. The term *spectral* refers to the timbral content of sound — the precise mixture of partials (frequencies, amplitudes and initial phases) that make up the sound's waveform (see Chapters 9 and 13 for a discussion of sound and harmonics). Spectral composition is also a label that refers to a group of contemporary French composers (Dufourt, Grisey, Murail, Risset) that work extensively with spectral information. An overview of some of their compositional techniques can be found in "An Introduction to the Pitch Organization of French Spectral Music" by Francois Rose. Many other contemporary composers work extensively with spectral information as well, including John Chowning, Barry Truax and James Dashow in North America, and spectralist tendencies are clearly observable in the work of earlier twentieth century composers such as Andre Jolivet, Olivier Messiaen, Iannis Xenakis, Edgard Varese and Ben Johnston.

Because spectral techniques are based on the physical properties of sound, computer analysis and synthesis figure strongly in this type of composition. There are two broad approaches to composing with spectral information. The first method employs an analysis/resynthesis strategy — a specific source waveform is analyzed by computer for its harmonic content and this spectral information is then "resynthesized" back into the composition in ways the composer works out. The second approach to spectral composition utilizes *sound synthesis algorithms* to synthesize spectral information directly. We will introduce both strategies in this chapter.

## Spectral Analysis

Spectral analysis means to deconstruct a sound (a source recording or real-time audio signal) into its constituent partials. Analysis methods vary according to the content of the waveform, whether it is periodic or aperiodic in nature. One particular method, called the *Fast Fourier Transform* (FFT) lies at the heart of many computer programs that analyze complex periodic waveforms. The French mathematician Jean Fourier (1768-1830) discovered that any periodic wave can be described as a sum of individual sine waves, where each sine wave is described by its frequency, amplitude and initial phase. This suggests that there are two different ways to look at a sound: as a single pressure wave, or as a collection of individual partials. A *time domain* representation of a waveform displays the aggregated wave's changing amplitude (pressure) over time. A *frequency domain* representation plots the changing amplitudes of the individual partials over frequency (Figure 23-1).

Figure 23-1. Time and frequency domain representations of a 220 Hz Clarinet tone.



- clarinet.mp3 Clarinet playing A3 (220 Hz).

The FFT algorithm provides an efficient bridge from the time domain to the frequency domain representation of a wave. An FFT analysis proceeds by first "chopping up" the input signal into short time segments called *windows*. By performing the analysis over a whole series of windows the FFT program can capture the dynamic characteristics of a spectrum as it changes over time. Each window in the analysis defines a snapshot of the spectrum to be analyzed. The mechanics of the FFT analysis within a single window are quite complex. Suffice it to say that the FFT acts like a bank of filters — it measures the amount of energy in frequency “bins” that are equally distributed over the entire range of frequencies. This means that the bins of a given FFT analysis should match, or line up, with the spectral components in the input signal and most analysis programs provide ways to tune the FFT analysis to the specific sound being analyzed. The output from the FFT analysis are *frames* of data that contain the amplitude and initial phase values for each frequency bin. This information can then be displayed, resynthesized, or saved to data files.

The important thing to grasp here is that — while the time and frequency domains are really just different perspectives on the same thing — the frequency domain representation is particularly useful to the spectral composer because it deconstructs the wave into its discrete components that can then be used to inform compositional decision making. For example, one can see quite clearly from the frequency domain plot in Figure 23-1 that odd numbered harmonics contain much more spectral energy than do the even harmonics. This acoustic information could be reflected by the composer's preference for using the proportions 1:3:5:7 to determine critical aspects of the composition.

Some FFT programs allow the results from the sound analysis to be exported as data files containing the data in each frame of the analysis. Example (Example 23-1) shows what data from one frame might look like.

Example 23-1. Spectral snapshot of Clarinet tone in Figure 1.

```
(define clarinet
  '(1.00 .0842  2.01 .0017  3.00 .0420  4.01 .0029
    5.00 .0251  6.02 .0041  6.99 .0115  7.96 .0006
    8.99 .0032  9.98 .0003  11.00 .0018 11.96 .0003
    12.96 .0006 14.00 .0001 14.97 .0001 15.97 .0005
    17.95 .0002 18.96 .0001 19.07 .0001 22.06 .0001))
```

Example 23-1 displays the spectral information output by the FFT analysis of the clarinet tone in shown Figure 23-1. The information contains the first twenty harmonics from the analysis. The data are interpreted pairwise: the first number in each pair is the frequency ratio of a harmonic and the second number is its amplitude. Phase information is not represented here. The example was taken from the file "spectra.clm" from the Common Lisp Music source distribution. This file contains a large number of spectral sets that can be used as compositional source material without having to actually perform an FFT analysis yourself.

## Spectral Resynthesis

Output from an FFT analysis is simply data. As such it can be mapped, manipulated and processed in various ways to resynthesize the information back into a composition. The nature of this transformation is, of course, up to each composer's imagination (see Chapter 16 for a general discussion of mapping). In the following example we define a simple set of tools that allow spectral information formatted as harmonic and amplitude pairs such as those shown in Example 23-1 to be reshaped into different tonal gestures.

Example 23-2. Spectral processing tools.

```
(define (spectrum-maxfreq spec)
  ;; return freq of last harmonic in spec
  (list-ref spec (- (length spec) 2)))

(define (spectrum-maxamp spec)
  ;; iterate successive tails of spec and find
  ;; the max amplitude
  (loop for tail = spec then (rest (rest tail))
        until (null? tail)
        maximize (second tail)))

(define (spectrum-freqs spec)
  ;; return only the freq components of spectrum
  (loop for tail = spec then (rest (rest tail))
        until (null? tail)
        collect (car tail)))

(define (spectrum-length spec)
  ;; return number of partials in spec
  (/ (length spec) 2))
```

The **`spectrum-maxfreq`** function returns the highest harmonic scaler in *spec*. Since the harmonic pairs in *spec* are ordered lowest to highest the function simply returns the second-to-last element in the list, which is the frequency value of the highest harmonic. The **`spectrum-maxamp`** function returns the loudest amplitude in *spec*. Since there is no way to predict which harmonic in a spectrum will be the loudest the function must search the entire spectral list for the maximum amplitude. The *tail* variable holds successively shorter versions of *spec*. On the first iteration *tail* is equal to **`spec`** but each iteration thereafter sets *tail* to **`(rest (rest tail))`** — the *remainder* of the list beginning at the next pair of elements to the right. As *tail* moves successively rightward though *spec*, the form **`(second tail)`** always accesses the amplitude value of the current pair at the front of the list. Iteration continues until *tail* is empty and the predicate **`null?`** returns true, at which point the **`until`** clause stops the iteration. The **`spectrum-freqs`** function iterates *spec* in a manner identical to *spectrum-maxamp* but collects **`(first tail)`** — the frequency component of the current pair in *tail*, thus reducing the spectrum to a list of its frequency values.

Given this basic set of tools we can write a general mapping function to transform spectral information into different shapes:

Example 23-3. Rescaling spectral data.

```
(define (rescale-spectrum spec a1 a2 k1 k2)
  ;; rescale spectrum to lie between keynums k1 and k2
  ;; and amplitudes a1 to a2
  (let* ((fund (if k1 (* (hertz k1) (first spec)) 1.0))
         (key1 (keynum fund :hz))
         (key2 (keynum (* fund (spectrum-maxfreq spec)) :hz))
         (soft 0)
         (loud (spectrum-maxamp spec)))
    (loop for tail = spec then (rest (rest tail))
          until (null? tail)
          collect
          (list
           (rescale (keynum (* fund (first tail))
                            :hz)
                    key1 key2 (or k1 key1) (or k2 key2))
           (rescale (second tail) soft loud a1 a2)))))
```

The **`rescale-spectrum`** function rescales a spectrum such that it lies proportionally between the specified key numbers *k1* and *k2* and the amplitude values *a1* and *a2*. If *k1* or *k2* is false their values default to the lowest and highest frequency values in *spec*. If *k2* is less than *k1* then the spectrum will be inverted. The spectrum returned by **`rescale-spectrum`** is formatted as a list of lists rather than as list of paired values. Reformatting the spectral data allows each partial in the spectrum to be represented explicitly as a sublist containing a frequency and an amplitude value. This reduces the total length of th spectrum by half, and allows that each partial to be processed atomically inside patterns or lists. Given this simple spectral toolkit we can now define a musical process that creates tonal gestures from spectral information. The main process **`spectral-zither`** creates zither-like material from the clarinet spectrum using two “helper” processes. The first helper plays an upward or downward strum of a rescaled spectrum and the second produces random plucks of the same material:

Example 23-4. Spectral gestures.

```
(define (strum-it play rate dur amp up?)
  ;; play ascending or descending spectrum
  (process for s in (if up? play (reverse play))
           output
           (new midi :time (now)
                :keynum (first s)
                :amplitude amp
                :duration (vary dur .1))
           wait rate))

(define (pluck-it play rate dur amp len)
  ;; play spectrum in random order
  (let ((pat (new heap :of play)))
    (process while (< (now) len)
             for s = (next pat)
             output
             (new midi :time (now)
                  :keynum (first s)
                  :amplitude (interp (now) 0 amp len (/ amp 2))
                  :duration (odds .2 (* dur 4)
                                  (vary dur .1)))
             wait rate)))
```

The main process **`spectral-zither`** takes a *spectrum* list (such as the clarinet defined earlier in this chapter) and generates a self-similar series of spectra in which each rescaled version is transposed to a specific harmonic level in the original spectrum. The amplitudes of each partial in the original determines the proportional length of each gesture — the louder the partial in the original spectrum the longer the transposed gesture lasts. Each gesture consists of two parts, a loud initial strum followed by random plucking of the same material. The amplitudes of the random plucks diminish over the length of each gesture.

Example 23-5. The spectral zither process.

```
(define (spectral-zither spectrum key1 key2 tim1 tim2)
  ;; transpose tonal gestures to successively higher
  ;; positions in spectrum. duration of each gesture is
  ;; proportional to amplitude to the partial it is
  ;; based on
  (let ((spec (rescale-spectrum spectrum
                                key1 key2 tim1 tim2)))
    (process for x in spec
             for k = (first x)
             for a = (second x)
             for topk = (between 100 113)
             for play = (rescale-spectrum spectrum k topk .4 .9)
             for up? = true then (odds .5)
             sprout (strum-it play .075 2.3 .9 up?)
             sprout (pluck-it play (vary .075 .2) .6 .9 a)
             wait a)))
```

We now generate both upward and downward versions of the spectral zither. The second version uses channel tuning to approximate the spectral microtones.

Interaction 23-1. Playing the zither.

```
cm> (events (spectral-zither clarinet 10 80 2 6)
            "zither.midi")
"zither-1.midi"
cm> (events (spectral-zither clarinet 100 20 4 3)
            "zither.midi" :channel-tuning 9)
"zither-2.midi"
cm>
```

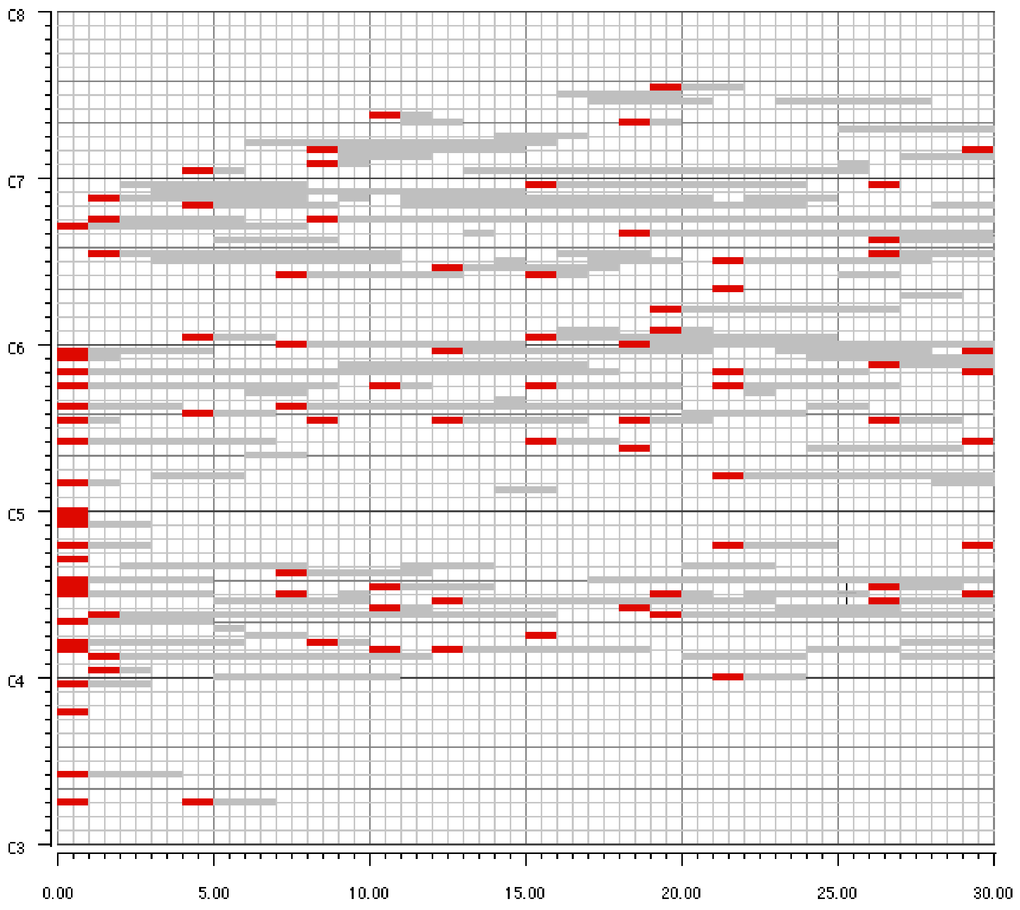
- zither-1.midi
- zither-2.midi 8 channels, microtones.
- clearbends.midi Clear pitch bends.

## 1706°F

To conclude this section on analysis and resynthesis methods we examine a spectral composition that was composed with information derived from the FFT analysis of a tam-tam tone. The piece is *1706°F*, for chamber ensemble and electronic sounds, written by the composer Michael Klingbeil. The analysis data for the composition was first organized in Common Music and then manipulated by the composer to create the instrumental parts and synthetic sounds events (samples) which are triggered at appropriate times during the performance of the piece.

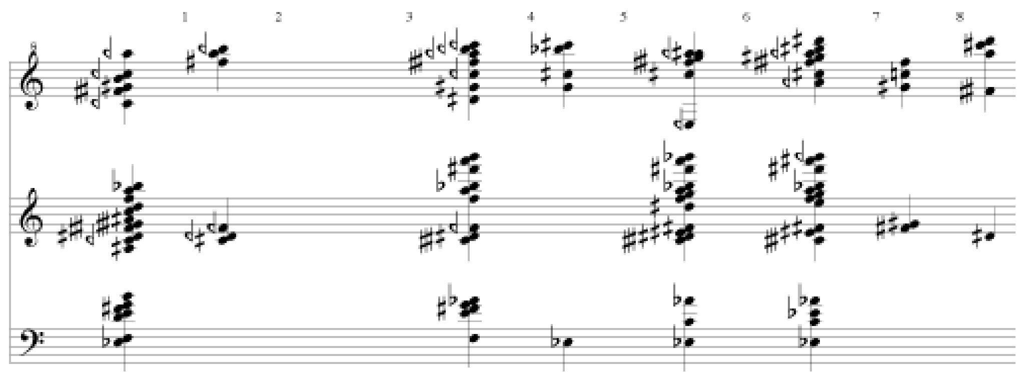
The original tam-tam sound was approximately 11 seconds long which resulted in over 300 FFT windows, each corresponding to about 35 milliseconds of actual sound. Rather than try to recreate the tam-tam sound with the instrumental sounds, the composer chose to conceive of each analysis window as a kind of “chord”.

Figure 23-2. Plot of the first 30 analysis windows where the horizontal axis corresponds to window number.



Common Music was used to manipulate the individual chords, first by finding common tones between each successive chord and then by selecting chords based on the number of non-common tones with its predecessor. These selected chords are highlighted in the preceding graphic. The remaining pitches, shown in light gray, became a structural background layer which was used in part as a basis for the electronic sounds. The melodic and contrapuntal contours of the instrumental parts were then created by choosing from the reservoir of pitches available in each chord. The compositional method was not always strictly algorithmic: at times pitches were selected from a larger corpus (such as all of the pitches found in a particular time window or all the pitches sounding within a particular time region spanning across several windows).

Figure 23-3. Musical notation of 8 chords selected from the first 9 analysis windows. (Note that the pitches may differ slightly from the graph shown above since some chords were selected using a different method of quantizing frequencies to tempered pitches.)



Selected chords were translated into common practice notation (with quarter tones) to facilitate the free composition of the instrumental parts. In some cases a chord may correspond to only a few seconds of composed music, or in other instances as many as 30 or 40 seconds.

Simple processes are used to generate the electronic sounds. The following is a MIDI simulation of electronic events numbered 40 and 41 in the score. Event 40 simply sounds the pitches of time window 5 in an ascending arpeggio. The pitches are doubled at the octave, the octave plus a fifth, and at two octaves (this is to better simulate the electronic sounds which are actually used in the piece). Event 41 is a chord which sounds all pitches of analysis window 6 simultaneously.

In this example a portion of the data used in the piece is bound to the variable *tam-tam-data*. Each element of the list corresponds to the spectrum of a single analysis window (the first five windows are empty for this example).

Example 23-6. The tam-tam FFT data.

```
(define tam-tam-data
  '(() () () () ()
    (156.77036815728735 90.637619 259.9496481697027 82.718292
     286.574546692396 100.874054 306.25563026238274 93.336433
     324.7524389568296 100.055481 337.44764076690535 102.149223
     364.6634271878465 230.778137 415.7531705386655 119.252495
     606.0946243520343 86.700752 693.632505316461 99.007416
     793.6624934018535 122.259544 884.1881093270954 157.20726
     933.8675278134929 145.184113 1088.9671367780802 89.725365
     1488.6904580535806 101.671417 1511.527861059107 104.7155
     1618.8507000903057 84.252243 1713.9968634651475 112.454514
     1765.292105410349 89.296089 1890.2215478184416 99.125633
     1908.0848686296388 92.404488 1999.0901910667542 99.439217
     2041.0190392835411 105.710556 2179.3234227680086 88.835609)
    (156.9226656365204 84.869324 260.82045952862836 90.014053
     285.1896751834164 111.457977 307.17019597206075 99.64637
     336.09609235859693 120.298607 362.75665648480026 225.252136
     415.73593078208086 115.339218 655.5235086291844 87.095612
     692.9686610934529 82.759056 793.4209536257428 108.587471
     850.8079688652781 95.086945 883.5759843738408 150.945786
     932.8432789368185 150.02179 1088.3966109501614 85.69706
     1490.71509379789 112.47464 1512.3888337014348 92.084824
     1619.7171692738077 82.596962 1712.6721745280083 102.734337
     1890.1368493016446 97.024925 1908.251766584189 94.600105
     1998.8849421756013 106.950005 2009.4111265591894 83.189438
     2041.6380351691066 102.523079 2402.793452296438 84.39566)))
```

As with the clarinet data, each window in the tam-tam data is organized as frequency and amplitude pairs. In the case of the tam-tam data the frequency for each harmonic is recorded in Hertz rather than in scaler form. Given our spectral tool kit and this tam-tam data we can define two sample chords from the piece:

Example 23-7. Two sample chord events.

```
(define (event40)
  (let ((data (rescale-spectrum
               (list-ref tam-tam-data 5) 0.5 1.0 #f #f)))
    (process for delta in (append (explsegs 3 0.5 4)
                                  (explsegs 4 0.6 1/2)
                                  (explsegs (- (length data) 7)
                                            2 1/8))
             for s in data
             for count from 0
```

```
            each kt in '(0 12 19 24)
            output (new midi :time (now)
                        :keynum (+ (first s) kt)
                        :amplitude (second s)
                        :duration 1.5)
            wait delta)))

(define (event41)
  (let ((data (rescale-spectrum
               (list-ref tam-tam-data 6) 0.7 1.0 #f #f)))
    (process for s in data
            output (new midi :time (now)
                        :keynum (first s)
                        :amplitude (second s)
                        :duration (between 8.0 10.0))
            wait 0)))
```

We now generate both chords together but sligtly offset in time.

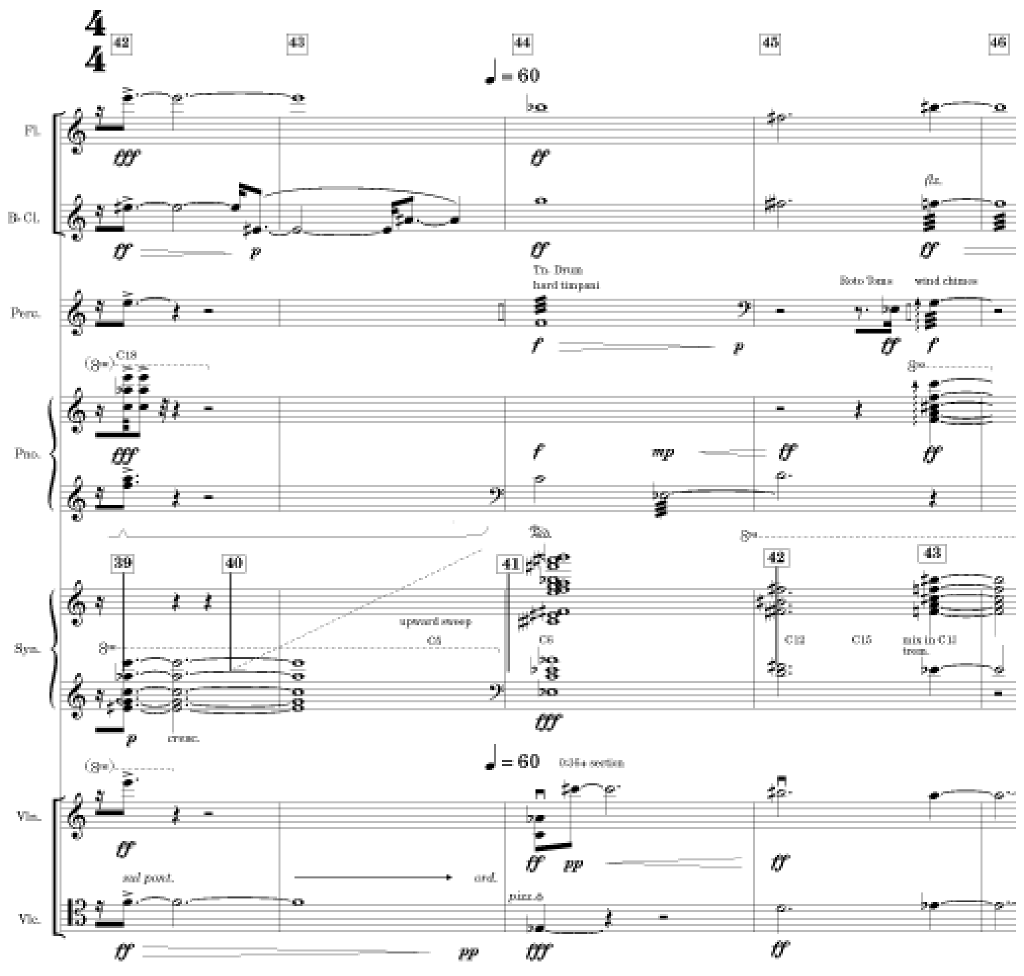
Interaction 23-2. Generating sample chords from 1706F.

```
cm> (events (list (event40) (event41))
            "1706Example.midi"
            '(0 3.75))
"1706Example.midi"
cm>
```

- 1706Example.midi

In the actual piece each pitch of the arpeggio is composed of a simple synthetic tone and a snippet of a tam-tam sound which is comb filtered to sound at the desired pitch. The file 1706Event40.mp3 contains the synthesized sound for Event 40. The chord of event 41 consists of noise filtered with very sharp filters at each frequency of the chord can be heard in the file 1706Event41.mp3. The upper pitches are doubled with a sampled crotale sound and a snippet of a cymbal sound is mixed in. Figure 23-4 shows the electronic part as it looks in the score.

Figure 23-4. This excerpt shows the combination of electronic events with the instrumental parts.



- 1706Example.mp3 Passage in performance.

## Spectral Synthesis

An alternate way to work with spectral information is to generate it directly using *synthesis algorithms* rather than through an analysis of specific sounds. A synthesis algorithm is a software program that calculates a waveform. As a very simple example of this process we will synthesize a spectrum consisting of the *combination tones* of two sine waves. When two sine waves *f1* and *f2* sound together, nonlinearity in the frequency response of our ears causes us to perceive frequency components above and below two sine waves. *Sum tones* are tones that lie at *f1+f2* Hz and *difference tones* lie at *f1-f2* Hz. Since sum tones are above the generating sine waves they tend to be masked but the difference tones can be clearly heard in the file diff.mp3. This sound example contains four difference tones, each difference tone is first “previewed” and then generated by pairs of sinewaves.

Example 23-8. Computing sum and difference notes.

```
(define (combo-notes n1 n2)
  (if (scale= n1 n2)
    '()
    (let ((f1 (hertz n1 ))
          (f2 (hertz n2)))
      (list (note (abs (- f1 f2)) :hz)
            (note (+ f1 f2) :hz)))))
```

The **combo-notes** function synthesizes combination notes by adding and subtracting two notes. If both notes are equivalent in the standard scale then the empty list is returned, otherwise the difference and sum in Hertz is calculated and the nearest notes to those frequencies are returned in a list. Since Hertz values are added and subtracted neither input *n1* or *n2* will appear in the output spectrum and the synthesized components will usually be inharmonically related to the original tones.

Interaction 23-3. Calculating sum and difference sets.

```
cm> (combo-notes 'd5 'e5)
(d2 ef6)
cm> (combo-notes 'a4 'ds5)
(fs3 c6)
cm>
```

In the audio domain this technique is referred to as *ring modulation* and the sum and difference tones are called spectral *side bands*. Ring modulation can be performed on individual pairs of frequencies (sine waves) or on complex waves. In the latter case, each frequency in the first wave combines with each frequency in the second to form an output spectrum that represents the "cross product" of the two inputs. We can implement compositional ring modulation using **loop** to append results returned by **combo-notes**:

Example 23-9. Synthesizing note sets by ring modulation.

```
(define (ringmod set1 set2)
  (scale-order (loop for n1 in set1
                     append
                     (loop for n2 in set2
                           append
                           (combo-notes n1 n2)))))

(define (play-rings notes rate)
  (process repeat 25
           for k = 70 then (drunk k 3)
           for r = (ringmod (list k) notes)
           for l = (length r)
           output
           (new midi :time (now)
                :duration (* rate l)
                :keynum k :amplitude .6)
           each s in r as i from 1
           output (new midi
                     :time (+ (now) (* rate i))
                     :keynum s
                     :amplitude .3)
           wait (* rate l)))
```

The **ringmod** function returns the spectral cross product of two input note sets. The Common Music function **scale-order** ensures that the notes in the synthesized spectrum are ordered low to high. See Chapter 9 for a discussion of **scale-order**.

Interaction 23-4. Ring modulated melodic line.

```
cm> (events (play-rings '(c4 e4 g4) .1)
            "rings.midi")
"rings-1.midi"
cm>
```

- rings-1.midi

## Frequency Modulation

In the remainder of this chapter we will examine *Frequency Modulation* (FM) as a way to synthesize tonal material for composition. FM audio synthesis was discovered by my teacher John Chowning at CCRMA, Stanford University. Chowning's FM paper The Synthesis of Complex Audio Spectra by Means of Frequency Modulation is a seminal paper in Computer Music and readers who are interested in the technique of FM should read it. Good non-technical discussions of FM audio synthesis are available in Computer Music by Dodge/Jerse and in Road's The Computer Music Tutorial William Schottstaedt's An Introduction to FM included in the CLM source distribution discusses the theory behind FM and also shows the reader how to design FM instruments in Common Lisp Music. Schottstaedt's FM instrument **`fm-violin`** is briefly discussed in Chapter 24. A number of composers, including John Chowning, have used FM as a compositional technique as well as for audio synthesis. FM is a particularly powerful synthesis algorithm because it can describe many different types of spectra using only a few simple control parameters.

Anyone who has listened to a good violinist can understand the basic idea behind frequency modulation. One of the ways a trained violinist adds warmth and expression to sound is by adding a small amount of *vibrato* to the tone by rocking the string finger back and forth in a constant manner. When the tip of the string finger angles toward the violinist the string is shorted by a very small amount and the frequency of the tone rises slightly; when the finger rocks back in the other direction the string is lengthened and a slight drop in pitch is produced. The constantly changing string length has the effect of *modulating*, or altering, the frequency of the main tone by a small percentage on either side of its center frequency. Vibrato is a periodic oscillation with two components: a rate (frequency) and an amplitude. The rate of oscillation is how fast the violinist moves the finger back and forth and can be measured in Hertz, or cycles per second. The amplitude of the vibrato determines how much the main tone deviates around its center frequency. The term *amplitude* in this case is really describing an amount of change of frequency called *deviation*. Typical values for vibrato frequency lie around six Hertz with a deviation of about 20 cents, or 1 percent around the center tone. The effects of changing the rate and deviation of a periodic vibrato can be heard in vib-1.mp3. The example demonstrates vibrato rates of 2, 4 and 8 Hz with deviations of 1/2. 1 and 2 percent around a 440 Hz tone.

### FM Synthesis

What we have described so far is a system in which one frequency is slightly modulated by another. Now imagine replacing our violinist with a robot capable of producing vibrato at extreme (audio) rates and with very wide deviations. As the rate of vibrato increases and its deviation becomes wider about the center frequency, the carrier becomes distorted to such an extent and so quickly that modulation is no longer perceives as a changing central pitch but as a changing *timbre* of the wave. “Robot” vibrato rates of 20, 40 and 80 Hz with deviations of 10, 25, and 50 percent around a 440 Hz tone can be head in vib-2.mp3 This is FM synthesis. FM synthesis generates a complex waveform by *modulating*, or changing, one wave called the *carrier* by another wave, called the *modulator* at audio rates. This modulation causes spectral sidebands to appear at multiples of the modulator above and below the carrier:

carrier ± ($k$ * modulator)

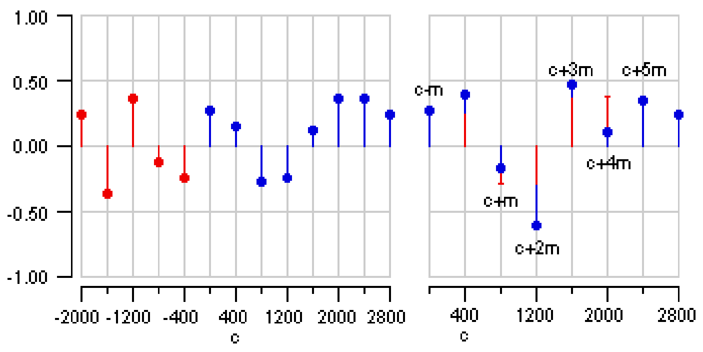
where $k$ is the order of the sideband, For example, a carrier of 400 Hz and a modulator of 100 Hz produces sidebands at the frequencies shown in Table 23-1.

Table 23-1. Sidebands arrangements for a carrier of 400 Hz and modulator of 100 Hz.

| c-3m | c-2m | c-1m | c | c+1m | c+2m | c+3m |
|---|---|---|---|---|---|---|
| 100 | 200 | 300 | 400 | 500 | 600 | 700 |

Note that at some value of *k* the lower sideband will become a negative frequency component. A negative frequency can be understood as a positive frequency with its phase inverted. When negative components wrap around to the positive domain their amplitudes invert and sum with any sideband located at that same spectral position (Figure 23-5).

Figure 23-5. Sideband arrangements for a carrier of 400 Hz and a modulator of 400 Hz. The first graph contains the negative frequency components, the second shows them inverted in the positive frequency domain.



- fm1.mp3 Carrier=400Hz, C:M=1:1, Index=4.

For a given carrier frequency *c* and modulator *m* the *c:m ratio* determines the precise arrangement of the spectral sidebands. The *c:m* ratio of 1:1 shown in Figure 23-5 produces sidebands at integer multiples of the carrier frequency — in this instance the FM spectrum mimics the harmonic series with a "fundamental" at the carrier frequency. The 1:1 ratio is the only *c:m* ratio which produces all harmonics with the carrier at the fundamental. All other ratios either produce a fundamental different from the carrier, or do not generate all harmonics, or generate sidebands that are not multiples of the carrier and so produce inharmonic spectra (Table 23-2).

Table 23-2. C:M sideband arrangements for a carrier of 400 Hz.

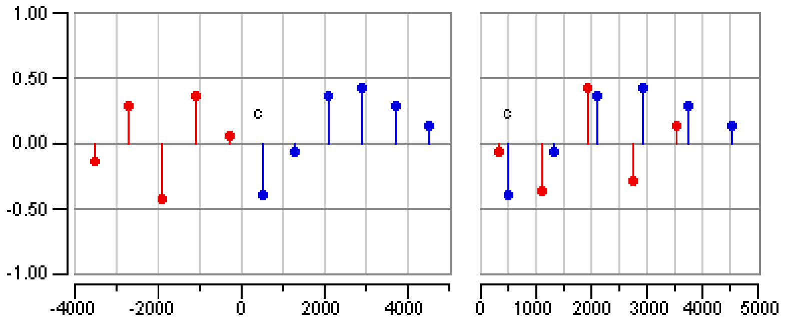
| c:m | sidebands | result |
|---|---|---|
| 1:2 | 400 1200 2000 2800 3600 | Odd harmonics. |
| 4:1 | 100 200 300 400 500 600 | fundamental not carrier. |
| 1:pi | 400 857 1657 2113 2913 3370 | inharmonic spectrum |

We can infer a few facts from Table 23-2:

- For the carrier frequency to be considered the fundamental, the *c:m* ratio must be 1:1 or *m* must be equal or greater than two times *c*.
- For ratios like 1:*m* where *m* is an integer greater than 1 the *mth* harmonic is missing from the spectrum.
- If *c* or *m* are not integers then inharmonic spectra result.

Figure 23-6 shows the sideband arrangements for a *c:m* ration of 1:*Phi* (the Golden Mean, or approximately 1.618): Notice that when negative inharmonic components wrap around to the positive domain they produce a "denser" spectrum because sidebands do not line up in the fashion seen in Figure 23-3.

Figure 23-6. Inharmonic spectra for a carrier of 500 Hz, c:m ratio of 1:phi and an FM index of 4.



- fm2.mp3 Carrier=400Hz, C:M=1:phi, Index=4.

For more information on C:M ratios and their effect on FM spectra see "Organizational Techniques for C:M Ratios is Frequency Modulation" by Barry Truax.

The precise relationship of sidebands to the carrier frequency depend not only on the rate of modulation but also the amount of deviation. Recall that deviation is the amplitude of the modulation — the maximum depth (in Hertz) that the modulation reaches on either side of the carrier. Deviation determines the number and amplitudes of the spectral components. Deviation is typically expressed indirectly, using an *FM index* that relates deviation to the modulation frequency:

deviation=index*modulator

If the FM index is zero then there is no deviation and only the carrier wave is heard. But as the index increases from zero an increasing amount of energy is “stolen” from the carrier and distributed over an expanding number of sidebands. (Figure 23-7) The precise amplitude relationships between the sidebands, carrier and FM index is determined by *Bessel functions* that describe what the amplitudes are in each sideband order for a given index. Bessel functions look a bit like dampened sinusoids, and can produce negative amplitude values. Sidebands with negative amplitudes are interpreted as having a positive amplitude but inverted phase.

In general, the number of sidebands *k* produced on either side of the carrier will equal the closest integer to the *FM Index* plus 1. Since there are two spectral components per side band, and the spectrum also contains the carrier, the total “density” of the spectrum will be approximately 2*k*+1. For example, an index of 2.5 will produce nine components: four side bands (eight spectral components) plus the carrier. However, since negative frequencies wrap around they may combine or cancel existing positive frequency components so the resulting spectrum many actually have fewer components.

## Composing with Frequency Modulation

Given a basic understanding of Frequency Modulation we can begin to work with FM synthesis as a compositional tool for computing both harmonic and inharmonic sets from three simple control values:

- a *carrier frequency*, the central tone for the note set.
- a *modulation ratio* that determines harmonicity of the resulting spectrum.
- An *fm index* that controls the number of sidebands, or the density of the spectrum.

[Function]

(**fm-spectrum** *carrier mratio index* [*spectrum*] [:unique *b*] [:invert *b*])

The **fm-spectrum** function returns the FM sidebands (frequencies and optionally amplitude values) calculated for a specified *carrier*, *mratio* and *index*. The types of values returned in the spectrum are determined by the value of the *spectrum* argument. If *spectrum* is **`:raw`**, then the both frequency and amplitude values are returned for each sideband and the spectrum may include negative frequencies. If *spectrum* is *:hertz* (the default) then negative sidebands are reflected into the positive frequency domain with their phases inverted. If **`:spectrum`** is **`:keynum`** or **`:note`** then sidebands are returned as key numbers or rounded to their nearest note names. The value of **`:spectrum`** can also be a list like **`(c4 c6)`** or **`(48 80)`** in which case the two values specify *boundaries* within which the spectral components will be fit regardless of which octaves the sidebands were actually located in. If note names are specified as boundaries then the spectrum is returned as notes otherwise key numbers are returned. In either case, the value of the **`:unique`** argument is used to determine if separate sidebands that produce identical note values are reduced to a single entry or not. The default value is true. If the **`:invert`** keyword is true then the spectrum is inverted before it is returned. If **`:unique`** is true then no duplicate key numbers or notes are returned for partials that map to the same spectral positions.

Interaction 23-5. Generating note sets.

```
cm> (fm-spectrum 220 1 1)
(220.0 440.0 660.0)
cm> (fm-spectrum 220 1 5 :note)
(a3 a4 e5 a5 cs6 e6 g6)
cm> (fm-spectrum (hertz 'c3) 2 5 :note)
(c3 g4 e5 bf5 d6 fs6 fs6 af6)
cm> (fm-spectrum 220 1 2 :keynum :invert t)
(33.0 37.980453 45.0 57.0)
cm> (fm-spectrum 220 1 9 '(a3 a5))
(c4 d4 e4 fs4 g4 bf4 c5)
cm> (fm-spectrum (hertz 'c3) 1.618 6 '(c4 g5) )
(c4 d4 e4 f4 g4 bf4 b4 cs5 d5 e5 f5 g5)
cm>
```

The following example uses **`fm-spectrum`** to generate a piano “improvisation”.

Example 23-10. Playing FM spectra.

```
(define (fm-harmony note )
  (let ((mrat (between 1.0 2.0))
        (indx 1))
    (fm-spectrum (hertz note) mrat indx
                 (list (transpose note -12)
                       (transpose note 12)))))

(define (fm-piano )
  (let* ((rhy (new random
                :of `((q weight 2)
                      (tq :min 3)
                      (,(new cycle :of '(e q e))))))
    (process for n = 60 then (drunk n 3 :low 40 :high 80
                                    :mode :jump)
             repeat 40
             for s = (fm-harmony (note n))
             for r = (rhythm (next rhy) 70)
             do (print (list (note n) s))
             output (new midi :time (now)
                         :keynum n
                         :amplitude .75
                         :duration (* r .9))
             output
             (new midi :time (+ (now) (rhythm 'e 70))
                  :keynum (list-ref s (pick 0 (1- (length s))))
                  :amplitude .7
                  :duration (* r .9))
             if (odds .6)
             each k in s
             output
             (new midi :time (now)
                  :keynum k
                  :duration (* r .9)
                  :amplitude (between .4 .55))
             wait r )))
```

The **`fm-harmony`** function returns an FM note set for a specified center note. The modulation ratio *mrat* is randomly selected between 1.0 (harmonic series) less than 2.0, so inharmonic spectra generally result. The *index* 1 produces 2 side bands on either side of the carrier for a total of 5 notes in each spectrum. But since the *spectrum* argument is specified as octave boundaries on either side of the carrier, fewer notes will be returned if transposed sidebands produce the same value inside the boundaries.

The **`fm-piano`** process plays 50 spectra generated by **`fm-harmony`**. On each iteration the process plays the center tone and either the lowest or highest note in the spectrum. Sixty percent of the time the process also adds the entire spectrum as a chord.

Interaction 23-6. Listening to fm-piano.

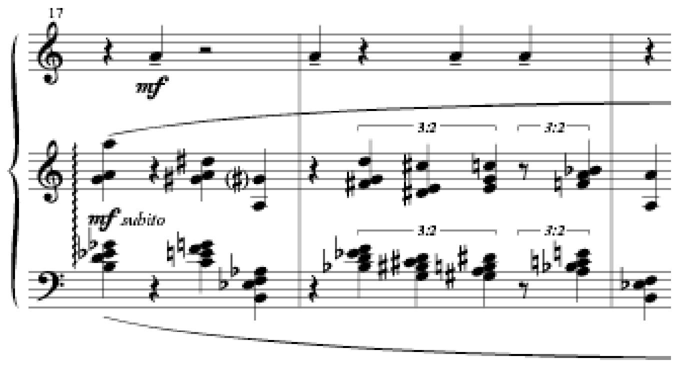
```
cm> (events (fm-piano ) "fm-piano.midi")
"fm-piano-1.midi"
cm>
```

- fm-piano-1.midi

## The Aeolian Harp

We conclude this chapter with two examples of FM tonal synthesis in *The Aeolian Harp*, a composition for piano and computer generated tape written by the author. Links to the full score, program notes and a recording are included at the end of this section. All the sounds in this 22 minute composition have their basis in either FM synthesis or in the justly tuned Aeolian scale. The tape music for Aeolian Harp consists of real sounds (primarily wind and plucked piano strings) that have been transformed or colored in some manner appropriate to the composition. These techniques include resonating comb filters, time stretching, and cross synthesis. Cross synthesis was used to create the “singing wind” that occurs toward the end of the composition as wind sounds take on the spectral properties of the Buddhist chant Zen Sho Kada sung by the Rev. Haruyoshi Kusada of Berkeley, California. A short passage from the seventh *Visions Fugitive* by Serge Prokofieff (title Arpa) appears briefly with Zen Sho Kada. The tape part was created using Common Lisp Music, a sound synthesis language developed by William Schottstaedt at CCRMA, Stanford University. All of the melodic and harmonic structures in the piano part were derived through Frequency Modulation. Many of the piano sections follow a similar technique: basic carrier, mratio, and FM index values are first determined using envelopes or patterns and then varied by some slight random amount to produce fresh spectra on each output set. The first example (Example 23-11) comes from the “bell section” near the beginning of the composition. The piano plays a constant A440 pulse with a harmonic accompaniment generated by FM using a carrier is A440, a mratio and an index:

Figure 23-7. Center frequency of A4 or A3, mratio between 1.1 and 1.6, and FM index of 2.4.



- ah1.mp3 Piano and tape, measures 17-22 (page 2 of score).

Here is a simplified version of the process that generated the bell-like FM chords.

Example 23-11. FM Bell harmony.

```
(define (fm-bell-notes tone )
  (let ((cen (hertz tone))
        (rat (between 1.1 1.6))
        (ind 2.4))
    ;(loop for n in notes collect (- (degree n) tone))
    (fm-spectrum cen rat ind
                 (list (transpose tone -12)
                       (transpose tone 12)))))

(defun fm-bell (reps rate)
  (let ((tones (new random
                 :of '((a4 :weight 3)
                       (a3 :max 1)))))
    (process repeat reps
             for tone = (next tones)
             for bell = (fm-bell-notes tone)
             output (new midi :time (now)
                         :keynum tone
                         :duration 3
                         :amplitude .6)
             when (odds .65)

             each k in  bell
             output (new midi :time (now)
                         :keynum k
                         :duration 3
                         :amplitude .4)
             wait rate )))
```

In Interaction 23-7 we listen to twenty **`fm-bells`**.

Interaction 23-7. Listening to fm bells.

```
cm> (events (fm-bell 20 1.25) "fm-bell.midi")
"fm-bell-1.midi"
cm>
```

- fm-bell-1.midi

The second example from Aeolian Harp demonstrates melodic material generated through FM synthesis. The excerpt comes from the beginning of the long piano solo in the middle of the piece. The entire solo is generated by FM. Each measure in Figure 23-8 constitutes one FM spectrum. The measures are played at a constant rate no matter how may notes are in the spectral set — the larger the set the faster it must be played but the carrier is always heard as the first note of each measure. The FM index in this section was controlled as a function of time. Over the course of the section the index and note boundaries increase so that the sets become more jagged and chromatic. Figure 23-9 shows what the spectra look like near the end of this section.

Figure 23-8. Carrier A4, A3 or A5, mratio between 1.1 and pi, index between 2 and 2.5.

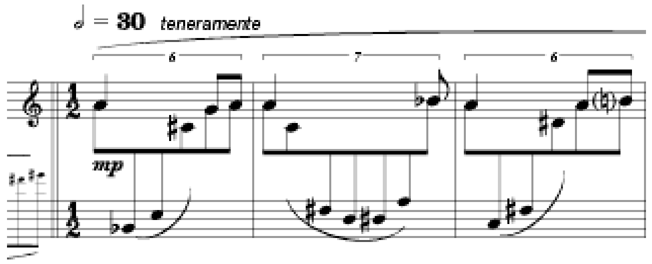
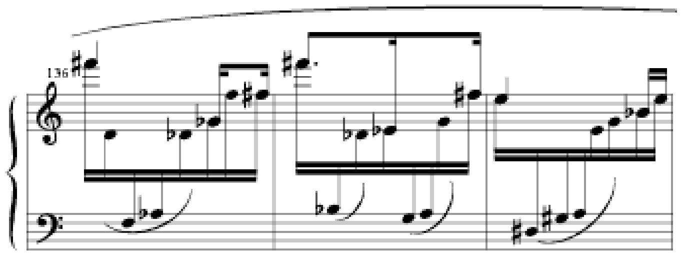


Figure 23-9. Same process with increased FM index and enlarged note boundaries.



- ah2.mp3 Piano solo, measures 118-150 of score.

The following process example is a slightly simplified version of the melodic generator.

Example 23-12. FM melodic generator.

```
(define (gen-mel cen rat ind low high)
  (fm-spectrum (hertz cen)
               rat ind
               (list (transpose cen low)
                     (transpose cen high))
               :scale-order nil
               :invert true :unique nil))

(define (fm-melody measure ratlow rathigh
                   indlow indhigh blow bhigh)
  (let ((starts '(a4 a4 a4 a3 a4 a4 a5 a4)))
    (process for cen in starts
             for rat = (between ratlow rathigh)
             for ind = (between indlow indhigh)
             for mel = (gen-mel cen rat ind blow bhigh)
             for rhy = (/ measure (+ 1 (length mel)))
             output (new midi :time (now)
                         :keynum cen
```

```
                  :amplitude .6
                  :duration .4)
      each n in mel as r from rhy by rhy
      output (new midi :time (+ (now) r)
                  :keynum n
                  :amplitude .4
                  :duration 2)
      wait measure)))
```

The **`gen-mel`** function generates an inverted fm spectrum (the “undertones” of the carrier) fit between two boundary notes *low* and *high*. Since **`:unique`** is false, the spectrum is not reduced to a set after the spectral components have been fit between the boundaries. The **`fm-melody`** process plays a series of FM spectra generated by **`fm-mel`** as a single melodic line. Each spectrum represents one measure of music with a constant duration; the more notes in a spectrum the faster they are be played.

This example approximates the piano in Figure 23-8:

Interaction 23-8. Listening to fm-melody

```
cm> (events (fm-melody 1.8 1.1 pi 2.0 2.5 -24 12)
            "fm-melody.midi")
"fm-melody-1.midi"
cm>
```

- fm-melody-1.midi

The score, recording and program notes of the entire composition are located in this chapter's directory:

- Aeolian Harp Score
- Aeolian Harp Recording (recorded on April 3, 2002 at the University of Illinois; Mei-Fang Lin, pianist.)
- Aeolian Harp Program Notes

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file spectral.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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H. Taube
06 May 2003

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# 24 Beyond MIDI

All of the algorithms, examples and Etudes in this book have described musical sound in terms of MIDI event and MIDI score files. The simplicity of MIDI has been a convenience for us — a way to generate musically satisfying results while maintaining our focus on larger compositional issues and algorithmic design. Indeed, the MIDI protocol *is* very convenient: it is simple, universally available on all makes of computers and an industry standard that has been adopted by a multitude of software applications. But the benefits of the MIDI protocol — its simplicity and ease of use — come only at the price of limiting compositional expressiveness and control over the production of the acoustic signal. We have already encountered some of MIDI's limitations during the course of this book. For example, we have seen that MIDI has an extremely weak representation of frequency, which it defines in terms of only 128 “key numbers”. MIDI is biased towards the western equal tempered chromatic scale and has no notion of fequency spelling. The protocol has many other limitations as well, including an very slow data rate, very poor 7-bit quantization, and an extremely limited channel, or instrument, address space.

Perhaps the most severe restriction that MIDI imposes on a composer has more to do with its nature than its features: MIDI is a *control protocol* and not a *language* for sound synthesis. From the perspective of a composer using MIDI, a synthesizer is a “black box” whose hardwired operations are hidden behind the messages that MIDI transmits. MIDI makes it very easy to use the synthesizer but at the price of control over synthesis algorithms. One solution to this problem is to purchase a number of different synthesizers, each with its own distinct capabilities. It is quite common for composers who work seriously with MIDI to have a whole rack full of such expensive synthesizers, samplers and effects processors. But many composers eschew MIDI completely in favor of *sound synthesis languages* that permit the specification of audio signals on a sample-by-sample basis. These languages provide a substitute for MIDI syntheses that render sound from score or performance information using specific audio algorithms, or “digital instruments” the composer chooses to work with.

In this chapter we introduce sound synthesis as an alternative to MIDI sound generation. We will learn how to work with two different sound synthesis languages from Common Music: Csound and Common Lisp Music. Both of these languages exhibit a certain similarity in their design since they both provide the composer with “toolboxes” of audio operators and an ability to design audio instruments from scratch. Which language a composer chooses to work with is largely a matter of preference, or deciding which language features are most important for a given compositional project. Among the factors to consider when selecting a synthesis language are: availability on your computer and operating systems, speed of execution, interactive control, expressive power, extensibility of system, and how well the software is supported. Of course it is not possible in this chapter to explain much about sound synthesis generally or these synthesis languages in particular — these topics can only be properly addressed in books. *Computer Music* by Dodge/Jerse is an excellent introductory text that discusses most of the synthesis techniques used by composers today. Its presentation is simple and generic and can be applied any synthesis language you choose to work with.

## Sound Synthesis Languages

A sound synthesis language is a software system that implements audio processing operators called *unit generators* and provides a syntax for combining these operators together to form larger modules called *instruments*. A digital instrument is an algorithm that generates a digital audio signal. Each type of unit generator implements a distinct, generic operation on an audio signal. Common unit generators include oscillators, envelope generators, wave tables, delay lines and various types of digital filters. By routing the output of one unit generator into the input of another, a relatively small number of different generator types can be “chained together” in different configuration to implement different synthesis algorithms. Each collection, or network, of operators is called a synthesis instrument. Synthesis instruments are often depicted as a graph or a “patch” in which the different unit generators are connected by lines representing the signal's path as it flows from generator to generator.

## Synthesis runtime environments

All synthesis languages make a distinction between the activity of designing an instrument and actually using the instrument to create an output audio signal. This latter activity is the responsibility of the synthesis *runtime environment*. The job of the runtime environment is to route external synthesis data to the synthesis instruments defined inside the environment. This routing may be implemented in or out of real time. Some runtime environments allow synthesis data to be routed interactively, via input controllers or by moving sliders and other graphical interface controls. Other languages implement a *score file* interface to sound synthesis. A score file is a text file that contains a time ordered series of *instrument calls*, where each instrument call is roughly analogous to a note in a traditional music score. An instrument call in a score file associates the name, or identifier, of a synthesis instrument with *parameter data* that defines one particular sound. The parameter data provides the initial conditions for the instrument's audio algorithm as it generates sound output. The synthesis data enters the instrument through its *parameters*, or input variables. Thus the notion of an "instrument call" in a score file is very similar to a function call in Lisp, and as we will see shortly, a synthesis instrument can be implemented as a specialized type of Lisp function.

## Instruments and parameters

One of the challenges of working with digital instruments is to learn which parameters produce the interesting results. Since each parameter adds an additional dimension of complexity to a digital instrument, composers may spend hours experimenting with parameter values for an instrument in order to learn about its acoustic capabilities. The first task in this process is to understand what each parameter means to a given instrument, and what the valid range of values are for every parameter. Since each synthesis instrument implements its own synthesis algorithm, the number and types of parameters can vary widely from instrument to instrument. Very simple instruments may have only a few parameters but it is not uncommon for a synthesis instrument to have twenty or more parameters associated with it! An instrument might use a large number of parameters because it requires a large amount of input data (as in the case with additive synthesis instruments, for example) , or because the instrument supports many optional features that affect its behavior, for example vibrato, glissando, reverberation, and so on. In order to reduce the amount of parameter data, a language like CLM allows *default values* to be associated with parameters such that a value needs only be specified in an instrument call if it is actually different from the value the instrument would otherwise use. Although the order and meaning of instrument parameters cannot be known without consulting documentation, many instruments commonly adhere to a conventional interpretation of the first few parameters:

*name start duration frequency amplitude*

where *name* is the name or identifier of the instrument, *start* is its start time in the score in seconds, *duration* is its duration, *frequency* is in Hertz and *amplitude* is its amplitude value. The following table shows what two instrument calls might look like for Csound and Common Lisp Music.

Table 24-1. Example instrument calls in Csound and CLM.

| | |
|---|---|
| **Csound** | `i1 0 3 440 .5` |
| **CLM** | `(beep 33.2 10.0 8231.43 .75)` |

In the remainder of this chapter we will learn how to define new musical objects and use them in conjunction with the Csound and Common Lisp Music synthesis languages. Although our musical sounds will be rendered quite differently than using MIDI, it is important to realize that actually very little will change when viewed from the perspective of the metalevel. The only practical differences will be:

- Compositional algorithms will express sound using sound event classes that permit more explicit control over the syntheses of sound than using the MIDI event class.
- Each new sound event class has its own set of sound parameters, depending on the instrument it uses to render sound.
- Frequency will be specified using Hertz values rather than key numbers or note names.
- MIDI files will be replaced by Csound or CLM score files and audio files.

## Csound

Csound is a synthesis language that runs on almost every computer and operating system in existence today. As its name would suggest, Csound is written in the C programming language but the actual syntax of the synthesis language that it implements is not C. Csound's runtime environment can be controlled by gestural control or by input from score files. To generate sound, Csound combines what it calls an “orchestra” of synthesis instruments with synthesis data in a score files. Both the orchestra and score are simply text files with information encoded in csound's specialzed instrument and score syntax: an orchestra file has an ".orc" file extension and a score file has a ".sco" extensions. Although explaining the details of Csound's instrument syntax is beyond the scope of this book it is not difficult to understand its general behavior. We will start with the orchestra file first. Each synthesis instrument in an orchestra file is identified by a unique *instr*, or instrument number. Example 24-1 show the contents of of an orchestra file called "blur.orc" that contains a single instrument definition.

Example 24-1. Contents of the Csound orchestra file "blur.orc".

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1

;;;
;;; simple oscillator with envelope
;;;
;;; f1: 0  <size>  10  1 0 1 0 1 0 1 0 1 0 1

instr 1
idur = p3
iamp = p4*32767       ; scale logical amplitude
ifreq = p5
iattackt = p6
irelt   = p7

k1      linen   iamp, iattackt, idur, irelt
a1      oscili  k1, ifreq, 1
out     a1
endin
```

The instrument defined in Example 24-1 consists of a single oscillator controlled by an amplitude envelope. The first few lines of the file set the global sampling rate *sr*, control rate *kr* and number of audio output channels *nchnls* for sound synthesis. Our instrument's *instr* number is 1, which means that instrument calls that reference this instrument in the score file will use the name **`i1`** to refer to our instrument. The **`i1`** instrument requires 7 parameters; some of there parameters be seen in the example by their *p* names: (*p2*, *p3*, *p4* and so on). Table 24-2 lists the parameters of the **`i1`** instrument and shows example values for each parameters.

Table 24-2. i1 instrument parameters.

| name | time | dur | amp | freq | amp-attack | amp-release |
|---|---|---|---|---|---|---|

| **parameter** | p1 | p2 | p3 | p4 | p5 | p6 | p7 |
|---|---|---|---|---|---|---|---|
| **example** | i1 | 0.0 | 1.0 | .5 | 440 | .1 | .1 |

The next five lines in the instrument's definition initialize local variables used by the instrument. The instrument's duration *idur* is set to the value of the *p3* parameter. The instruments amplitude *iamp* is calculated by rescaling the logical amplitude in *p4* by 32767, which converts it into Csound's internal amplitude scale. The next three lines proceed in a similar way: the frequency *ifreq* in Hertz is set to the value of *p5*, the attack time of the instrument's amplitude envelope *iattackt* is set to *p6* and the envelope's decay time *irelt* is set to *p7*. The actual instrument patch appears in the next two lines; this is the runtime code that is executed repeatedly to generate the audio output signal. In the first line the envelope unit generator **`linen`** (line envelope) calculates the current sample's amplitude value *k1*, which becomes the amplitude input to an interpolating oscillator **`ocili`**. The oscillator produces a sine wave at the frequency *ifreq* and sets the variable *a1* to the current audio sample. The **`out`** statement then writes the current sample to the open audio output file.

## Defining a Csound event

Given the fact that we can define any number of instruments in Csound, each with its own set of parameters and sound characteristics, we must have some means for defining musical events at the metalevel that can output these sound descriptions to Csound scores. More specifically, in order to work with the **`i1`** instrument defined in the "blur.orc" file we must define a new musical event that will output **`i1`** instrument calls to a Csound ".sco" file.

The easiest way to create a new type of Csound event is to define it in terms of an already existing event class called **`i`**. Common Music provides the **`i`** event class so that different Csound instrument events can be created simply by *subclassing* this one event class. Recall from Chapter 11 that a subclass inherits all the slots and values declared by its *superclasses*. In the case of the Csound **`i`** class, its subclasses will automatically receive the following slot initializations:

:ins *integer*
  The instrument number of the csound instrument.
:time *number*
  The start time of the sound event.
:dur *number*
  The duration of the sound event.

We will use the **`defobject`** macro to define our new sound event class.

[Macro]
  (**`defobject`** *name* (*supers ....*) (*slots ...*) [(:parameters *...*)])

The **`defobject`** macro defines a new class of object. *name* is the name of the new class. Following *name* comes a list of zero or more superclasses from which the new class inherit slots and values. Following the superclass list comes a list of *slot specifications*. Each *slot* in the list is either the name of a slot or a list of the form:

  (*slot* [**`:initform`** *x*])

If an :initform expression is specified its value *x* becomes the value of *slot* if no initialization is provided when the object is created using the **`new`** macro. Following the slot specification list comes an optional parameter list:

  (**`:parameters`** *slot* ...)

This list declares the names of the slots that will produce parameter values for the instrument. The order of the slots in this list determines the order the values appear in the instrument call written to the score file. Example 24-2 shows the definition of a new class of object called **i1** that will output sound events for our **i1** instrument:

Example 24-2. Defining a new class of object.

```
(defobject i1 (i)
  (amp
   freq
   (attack :initform .05)
   (release :initform .25))
  (:parameters time dur amp freq attack release))
```

This **defobject** form defines a new sound event called **i1** that inherits slots and values from the **i** superclass. The defobject expression declares both slots and parameters for our new object. The **amp** and **freq** slots for **i1** are declared without specifying initial values, which means that these value will always have to be specified when an **i1** object is used. The attack and release slots, in contrast, are defined with initial values. If an initialization is not provided, the values for the attack and release slots will be .50 and .25, respectively.

Following the slot list comes the **:parameters** declaration list. The time and dur slots that appear in this list were inherited from the **i** superclass.

## Working with the i1 object

Once the **defobject** expression in Example 24-2 is evaluated we can create **i1** objects, as shown in Interaction 24-1:

Interaction 24-1. Creating i1 events.

```
cm> (new i1)
#i(i1 ins 1 attack .05 release 0.25)
cm> (new i1 :time 0 :freq (random 1000) :amp .5)
#i(i1 time 0 ins 1 amp 0.5 freq 693 attack 0.05 release 0.25)
cm>
```

## Generating a Csound score file

Now that the **i1** object has been defined we can write a simple example process that outputs **i1** events:

Example 24-3. Generate i1 events.

```
(define (blur )
  (process repeat 100
           output (new i1 :time (now)
                          :freq (between 20 2000) ; Hertz
                          :dur .4 :amp .1)
           wait .015))
```

At this point we can generate a Csound score. But before doing so we will first establish an output hook to automatically call the csound program with our orchestra file and score file whenever the **events** function outputs a csound score. You can define your own Csound hook function or use the **play-sco-file** hook defined by Common Music:

[Function]
  (**play-sco-file** *file* [:orchestra *s*] [:options *s*] [:output *s*] [:play *b*])

The play-sco-file functions plays the score *file* using the specified **:orchestra** file. **:options** is an optional string of commands that will be passed to the csound program when it starts. **:output** is the output sound file or "dac" if Csound will generate the audio signal directly to the speakers. By default, the **play-sco-file** function assumes that the csound program can be started using the shell command: "/usr/local/bin/csound". If this is not the location of the Csound program on your to reset the global variable ***csound*** to a different command string if your csound program is located in some other directory. If you do not have Csound installed on your computer you can generate the score file but you will not be able to run csound. In this case you should not set an output hook.

Before generating our score we will also define a csound *header* string that contains the definition of an amplitude envelope for the **linen** unit generator that **i1** uses. If a header string is specified to the events function Common Music will include the contents of the string in the top the output Csound score file. This string is not examined by Common Music and so can contain any text that is legal in a Csound score file. In our case the header string contains a csound *f statement* that defines an amplitude envelope. At this point we are ready to output events to a csound score file.

Interaction 24-2. Listening to blur.

```
cm> (set-sco-output-hook! #'play-sco-file)

cm> (define sco-header  ; define a score file header string
      "f1 0 8192 10 0.0  0.41 0.06 0.04 0.00 0.11 0.02 0.00 0.00")

cm> (events (blur) "blur.sco" 0
        :header sco-header
        :options "-d -m0 -A"
        :orchestra "blur.orc"
        :output "blur.aiff")
"blur.sco"
cm>
```

- blur.sco
- blur.mp3

Example 24-4 contains the first few lines of the "blur.sco" score file created in Interaction 24-2:

Example 24-4. Contents of blur.sco

```
; Common Music 2.4.0 output on 24 Mar 2003, 09:49:38
f1 0 8192 10 0.0  0.41 0.06 0.04 0.00 0.11 0.02 0.00 0.00
i1 0 0.4 0.1 300 0.05 0.25
i1 0.015 0.4 0.1 406 0.05 0.25
i1 0.03 0.4 0.1 975 0.05 0.25
...
```

Notice that the f statement from our header string appears before the first instrument call in the score file. After the score file was written our hook function automatically started the Csound program in "/usr/local/bin/csound" and passed it the command options "-d -m0 -A", our score file "blur.sco" and the orchestra file "blur.orc". Csound then generated audio output from the score to the file "blur.aiff".

# Common Lisp Music

Common Lisp Music (CLM) is a sound synthesis language developed by William Schottstaedt at CCRMA, Stanford University. CLM is fast, efficient, has real-time capabilities and runs on many different types of computers. Both CLM and Common Music share the same Lisp environment so, from the novice Lisp composers' perspective, CM and CLM may not appear to be separate software systems at all. Despite CLM's name, Common Lisp Music is actually implemented in Lisp and C. Lisp provides CLM an interactive development environment for designing and working with instruments. CLM uses C to implement the low-level unit generators and the runtime environment that performs the sample-by-sample calculation of the audio signal.

## Defining CLM instruments

Like Csound, CLM defines audio signals in terms of synthesis instruments acting on synthesis data. But in the case of CLM, both the audio instruments and the data they use are both represented in Lisp. CLM instruments are just specialized Lisp functions that generate audio signals. Instruments are defined using CLM's **definstrument** macro. A definstrument expression is normally placed in a Lisp source files with a ".ins" extension, and then compiled and loaded into Lisp. An ".ins" file can contain any number of CLM instruments and is in every way analogous to a Csound orchestra file.

For purposes of comparison, Example 24-5 shows the contents of an instrument file called beep.ins that contains a single CLM instrument definition. The instrument is called **beep** and it is equivalent to the **i1** Csound instrument defined earlier in this chapter.

Example 24-5. The contents of the CLM instrument file "beep.ins".

```
(definstrument beep (start dur freq amp amp-env)
  (let* ((beg (seconds->samples start))
         (end (seconds->samples (+ start dur)))
         (osc (make-oscil :frequency freq))
         (env (make-env :envelope amp-env :scaler amp
                        :duration dur)))
    (run
     (loop for i from beg to end
           do
           (outa i (* (env env) (oscil osc)))))))
```

Like **i1**, the CLM **beep** instrument consists of an oscillator and an amplitude envelope. The **let*** establishes local variables used in the calculation of the audio output signal. The variable *beg* is the first sample number to start generating in the audio output file and *end* is the ending sample. The *osc* variable holds an oscillator object and *env* is set to an envelope generator. The **run** expression establishes the scope of the instrument's "run time" statements: expressions inside the run block execute to generate a stream of audio samples. The runtime expression of **beep** consists of a loop that iterates over each sample number *i* and sets the variable *s* to the next sample output from the oscillator *osc* scaled by the output of the envelope generator *env*. The sample *s* is then written to the open output audio file.

## Compiling and loading instruments

Once a definstrument is defined the code must be complied and loaded into Lisp (Interaction 24-3). Compilation is a process that transforms program code (Lisp expressions) into equivalent low-level machine instructions. Compiled machine code executes many times faster than interpreted program code. Compilation is really a necessity for audio instruments due to the large amounts of data that they compute. For example, a stereo, CD-quality audio signal requires the calculation of 88,200 audio samples for each second of sound!

Rather than listen to the simple **beep** sound again we will take this opportunity to compile and load two instruments from the CLM software distribution. The main instrument is called **fm-violin** and it was developed by William Schottstaedt when he was a graduate student at CCRMA. The source code for the instrument is contained in the file v.ins located in the same directory as this HTML chapter file. We will not attempt to explain the instrument beyond describing it as an example of *cascade frequency modulation*, in which a carrier frequency is modulated by a complex wave from a cascade of three modulating oscillators. Despite its inherent complexity, the fm-violin instrument is actually very easy to work with. See chapter 23 for a general discussion of frequency modulation as a synthesis technique.

Interaction 24-3. Compiling and loading "v.ins".

```
cm> (cd "/Users/hkt/NM/24/")
"/Users/hkt/NM/24/"
cm> (compile-file "v.ins")
"/Users/hkt/NM/24/v.pfasl"
cm> (load "v")
"/Users/hkt/NM/24/v.pfasl"
cm> (compile-file "nrev.ins")
"/Users/hkt/NM/24/nrev.pfasl"
cm> (load "v")
"/Users/hkt/NM/24/nrev.pfasl"
cm>
```

Lisp Note:
The file extension for the compiled file will vary with each Lisp implementation.

In Interaction 24-3 the function **compile-file** is used to compile the Lisp program file containing the fm-violin instrument definition. The **load** function is then used to load the complied code into Lisp. It is possible to load interpreted program files as well as compiled files using the **load** function. In either case, the effect of loading is to evaluate the contents of the loaded file. We also compiled and loaded a reverberation instrument called **nrev** that we will use to add a "hall effect" to the audio output.

When the "v.ins" file is compiled and loaded both the fm-violin instrument as well as a sound event object for the instrument is automatically defined. This means that when you work with CLM instruments you do not have to use **defobject** to define a new object class. This process happens automatically by **definstrument**. The **fm-violin** instrument has almost forty parameters, but only the first four parameters must be specified in order to produce a sound. We can use Common Music's **object-parameters** function to inspect the list of parameters that the **fm-violin** contains.

[Function]
(**object-parameters** *object*)

Returns a list of sound parameters associated with the specified *object*.

Interaction 24-4. Listing an object's parameters.

```
cm> (object-parameters (new midi))
(time duration keynum amplitude channel)
cm> (object-parameters (new fm-violin))
(startime dur frequency amplitude fm-index amp-env
 periodic-vibrato-rate random-vibrato-rate
 periodic-vibrato-amplitude random-vibrato-amplitude
 noise-amount noise-freq ind-noise-freq ind-noise-amount
 amp-noise-freq amp-noise-amount gliss-env glissando-amount
 fm1-env fm2-env fm3-env fm1-rat fm2-rat fm3-rat fm1-index
 fm2-index fm3-index base frobber reverb-amount index-type
 degree distance degrees no-waveshaping denoise denoise-dur
 denoise-amp)
cm>
```

## The fm-morph process

One of the interesting properties of the fm-violin instrument is that it is capable of generating many different types of sound, not just violin timbres. The key to its flexibility lies, in part, with its many optional parameters. We will generate a sound file that demonstrates the effect of the *fm index* on the fm-violin's timbre. We start by defining a few envelopes that the instrument and process will use.

Example 24-6. Envelopes for the fm-violin example.

```
(define amp-env '(0 0 2 .5 3 .9 5 .6 75 .2 100 0))

(define morph-weight
  '(0 1 .2 10 .3 20 .35 50 .45 100 .5 100
    .55 100 .65 25 .97 20 .98 10 1 1))

(define spike-env
  '(0 .01 .2 .02 .3 .05 .35 .09
    .4 .1 .425 .15 .45 .2 .5 2.5
    .55 .2 .575 .15 .6 .1 .65 .09
    .7 .05 .8 .02 1. .01))

(define trough-env
  '(0 9 .2 8 .3 5 .35 4 .45 0 .5 0
    .65 4 .96 8 1 9))
```

The *amp-env* variable holds the amplitude envelope that the fm-violin will use. The other three envelopes are used by the process to calculate parameter values for fm-violin events. The *morph-weight* envelope is used to calculate the random weight of each note that the process generates as a function of time. The *spike-env* envelope controls the duration of each event, durations in the middle portion of the score will last significantly longer than at either the beginning or ending of the score. The *trough-env* envelope controls the *fm-index* of the process; its value is 9 at the outset and end of the process (quite large) and zero in the middle.

Example 24-7. The fm-morph process.

```
(define (fm-morph len rhy amp)
  (let* ((w1 1)
         (pat (new random
                :of `((c1 :weight .3)
                      (c4 :weight,(pval w1))
                      cs4 d4 ds4 e4 f4 fs4
                      (g4 :weight ,(pval w1))
                      gs4 a4 as4 b4
                      (c5 :weight ,(pval w1)))
                :for 1)))
    (process for i below len
             for pct = (/ i len)
             for frq = (next pat)
             for lo? = (scale= frq 'c1)
             do
             (set! w1 (interpl pct morph-weight))
             output
             (new fm-violin :startime (now)
                  :dur (if lo? 8
                           (interpl pct spike-env))
                  :amplitude amp
                  :frequency (hertz frq)
                  :amp-env amp-env
                  :reverb-amount .1
                  :fm-index (if lo? 2.5
                                (interpl pct trough-env)))
             wait rhy)))
```

## Generating an audio file

At this point we are ready to generate a CLM score. There are two possible score files that we can generate: a CLM score file or an audio file. A CLM score file is a text file with a ".clm" extension that contains a time ordered series of instrument calls. A CLM audio file contains the audio signal itself. We will take this opportunity to generate a stereo CD-quality audio file. Audio files can be generated from CLM in ".aiff", ".snd" or ".wav" formats simply by specifying one of these as the file extension in the **`events`** function. A CLM score has many possible initializations, only a few of the most basic are shown here:

:srate *number*
    The sampling rate of the audio file. The default is 22050.
:channels *number*
    The number of channels in the output audio file. The default is 1.
:trace *boolean*
    If true then each event will print its instrument and start time as it is output to the audio file.
:play *boolean*
    If true then CLM will play the audio file after it is written. The default value is true.

Consult the CLM documentation on **`with-sound`** for the complete list of audio file initializations. At this point we are ready to output our audio file:

Interaction 24-5. Generating audio output from fm-morph.

```
cm> (events (fm-morph 400 .1 .2) "morph.aiff" 0
            :srate 44100 :channels 2 :reverb 'nrev)
; File: "morph.aiff"
; Channels: 2
; Srate: 44100
; Reverb: NREV
"morph.aiff"
cm>
```

- morph.mp3

# Common Music Notation

We conclude this chapter with a brief discussion of Common Music Notation (CMN), also written by William Schottstaedt at CCRMA, Stanford University. Common Music Notation is a program that transforms Lisp expressions into a musical manuscript saved in an Encapsulated Postscript file (EPS). CMN has no editor and is not designed as a complete substitute for notation programs. However, it does an excellent job of music manuscripting and it is particularly useful for producing visual output from algorithms. The algorithms can simply use MIDI notes to represent the basic note data that CMN will display. While CMN's time-line and layout algorithms are very sophisticated, best results can be achieved by following two simple guidelines. First, if note spelling is important the keynum slot of the MIDI events should contain note names. If spelling is not important then keynum slot should contain Hertz frequency values. Second, to minimize problems with rhythm "quantization" of time values specified in seconds, express time values using exact ratios of a quarter note. This will be discussed in more detail in the next section.

## The CMN score

A CMN score is generated by specifying a file with an ".eps" or a ".cmn" file extension to the **`events`** function. In either case a EPS file containing the manuscript is generated. If a ".cmn" file is specified then a CMN input file will be saved along with the EPS output. A CMN input file is a text file that contains the Lisp expressions that generated the EPS manuscript. These expression can then be edited in any text editor to produce a more refined or detailed score. A CMN score supports many initializations, consult the CMN documentation for a complete listing. The most basic initialization are defined here:

:exact-rhythms *boolean*

> If true then the CMN score parses time values as exact rhythms. Exact CMN rhythms are ratios of the *quarter note*, not the whole note. For example, 1 means quarter note, 1/2 means eighth, 1/4 means 16th and so on. Using exact rhythms will avoid quantization errors that might otherwise occur if CMN parses note events containing time values specified in seconds.

:title *string*

> The title string to display in the output manuscript.

:size *number*

> The overall size of the score. The default value is 40; 24 produces a moderately small staff.

:staffing *list*

> A list of staff descriptions, see the next section for more information.

## CMN staff descriptions

A list of *staff descriptions* provides a way to determine the exact staff that each event will appear in when it is output by an algorithmic processes. The list of staff descriptions is specified to the **`:staffing`** initialization of a CMN score when it is created by the **`events`** function. Each staff description is a list in the format:

> (*id* [**`:name`** *s*] [**`:clef`** *sym*] [**`:meter`** *list*])

The first element *id* in the staff description is an integer identifier for the staff. Events output by a process are matched against all the staff ids in the score to determine which staff the object will appear in. In the case of MIDI events, the *channel* number of the MIDI event is used to specify this staffing value. The :name attribute is an optional string name for the staff. If a :name is not provided the name of the staff will be set to "staff-*n*" where *n* is the *id* of the staff. The value of the :clef attribute is an optional symbol clef name or a list of clef names. If a :clef value is not specified then CMN will use whatever clef it thinks best. If the clef value is :both then the grand staff is used. If the :clef value is a list of clefs it limits CMN's clef choices to the clefs mentioned in the list. The value :meter is an optional meter list for the staff, where the first element in the list is the number of beats per measure and the second is the measure beat. Example 24-8 shows the definition of a staffing list for a score with three staves of music:

Example 24-8. Example staff definitions.

```
(define trio-staves
  '((0 :name "Cello" :clef :bass :meter (4 4))
    (1 :name "Oboe" :clef :treble :meter (4 4))
    (2 :name "Flute" :clef :treble :meter (4 4))))
```

The staffing list in Example 24-8 species that any MIDI event with a channel value of 0 will appear in the staff named "Cello", channel 1 events will be written to the "Oboe" staff and channel 2 events will appear in the "Flute" staff.

## Creating a CMN manuscript

We first define a little example process that outputs MIDI events to our CMN score:

Example 24-9. defining a process to output CMN files.

```
(define duet-staves
  '((0 :name "Viola" :clef :alto :meter (4 4))
    (2 :name "Flute" :clef :treble :meter (4 4))))

(define (duet-cmn stf len nts)
  (let ((nts (new heap :notes nts))
        ;; choose quarter or two eighths
        (rhy (new random
               :of `(1 ,(new cycle :of '(1/2 1/2))))))
    (process while (< (now) len)
             for n = (next nts)
             for r = (next rhy)
             output (new midi :time (now)
                         :duration r
                         :keynum n
                         :channel stf) ; link to cmn staff
             wait r)))
```

Since our CMN example will involve exact rhythms the time values in the rhythm,m stream *rhy* are expressed using CMN rhythm ratios. Since these values are still proportional we could scale them by an arbitrary tempo factor if this process were to be used for both MIDI file and CMN file output.

Interaction 24-6. Outputting a CMN file.

```
cm> (events (list (duet-cmn 0 12 '(c3 d ef f g))
                  (duet-cmn 1 12 '(c5 d ef f g)))
            "duet.eps"
            :staffing duet-staves
            :size 24
            :title "Hiho!")
Manuscripting duet.eps...
"duet.eps"
cm> (events (list (ms-cmn 0 12 '(c3 d ef f g))
                  (ms-cmn 1 12 '(c5 d ef f g)))
            "duet.cmn"
            :staffing duet-staves
            :size 24
            :title "Hiho!")
"duet.cmn"
cm>
```

The file duet.pdf is a PDF version of duet.eps and the file duet.cmn contains the CMN input expressions that created it.

## Chapter Source Code

The source code to all of the examples and interactions in this chapter can be found in the file beyond.cm located in the same directory as the HTML file for this chapter. The source file can be edited in a text editor or evaluated inside the Common Music application.

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