Nyquist Reference Manual

Version 3.12


September 23, 2017

Carnegie Mellon University
School of Computer Science
Pittsburgh, PA 15213, U.S.A.
Contents

1 Introduction and Overview ...................................................... 10
  1.1 Installation ........................................................................ 10
    1.1.1 Troubleshooting .......................................................... 11
  1.2 Using NyquistIDE ............................................................. 11
  1.3 Using SAL ........................................................................ 12
  1.4 Command Line ................................................................... 12
  1.5 Helpful Hints ..................................................................... 13
  1.6 Using Lisp ......................................................................... 13
  1.7 Examples .......................................................................... 14
    1.7.1 Waveforms ................................................................. 14
    1.7.2 Wavetables ............................................................... 15
    1.7.3 Sequences ................................................................. 15
    1.7.4 Envelopes ................................................................. 16
    1.7.5 Piece-wise Linear Functions ......................................... 17
  1.8 Predefined Constants ........................................................ 18
  1.9 More Examples .................................................................. 19
2 The NyquistIDE Program .......................................................... 21
  2.1 NyquistIDE Overview ....................................................... 21
  2.2 The Button Bar .................................................................. 22
  2.3 Command Completion ....................................................... 23
  2.4 Browser ........................................................................... 23
  2.5 Envelope Editor .................................................................. 23
  2.6 Equalizer Editor ............................................................... 24
  2.7 UPIC Editor ....................................................................... 24
3 Behavioral Abstraction ............................................................. 27
  3.1 The Environment .............................................................. 27
  3.2 Sequential Behavior ........................................................ 29
  3.3 Simultaneous Behavior ...................................................... 29
  3.4 Sounds vs. Behaviors ....................................................... 30
  3.5 The At Transformation ..................................................... 31
  3.6 The Stretch Transformation ............................................... 31
  3.7 Nested Transformations .................................................... 32
  3.8 Defining Behaviors ........................................................ 32
6.2 Interoperability of SAL and XLISP ........................................... 63
   6.2.1 Function Calls .......................................................... 64
   6.2.2 Symbols and Functions ............................................... 64
   6.2.3 Playing Tricks On the SAL Compiler ................................. 64

7 Nyquist Functions ................................................................. 66
   7.1 Sounds ........................................................................... 66
       7.1.1 What is a Sound? ..................................................... 66
       7.1.2 Multichannel Sounds ............................................... 67
       7.1.3 Accessing and Creating Sound .................................... 67
       7.1.4 Miscellaneous Functions .......................................... 72
   7.2 Behaviors ....................................................................... 76
       7.2.1 Using Previously Created Sounds ................................. 76
       7.2.2 Sound Synthesis ....................................................... 76
           7.2.2.1 Oscillators ....................................................... 80
           7.2.2.2 Piece-wise Approximations ................................. 84
           7.2.2.3 Filter Behaviors ............................................... 87
           7.2.2.4 Effects .......................................................... 92
           7.2.2.5 Physical Models ............................................... 93
           7.2.2.6 Phase Vocoder .................................................. 96
           7.2.2.7 More Behaviors ............................................... 97
   7.3 Transformations ............................................................... 101
   7.4 Combination and Time Structure ....................................... 103
   7.5 Sound File Input and Output ............................................. 105
   7.6 Low-level Functions ....................................................... 112
       7.6.1 Creating Sounds ..................................................... 112
       7.6.2 Signal Operations ................................................... 114
       7.6.3 Filters ................................................................. 120
       7.6.4 Table-Lookup Oscillator Functions .............................. 124
       7.6.5 Phase Vocoder Functions .......................................... 126
       7.6.6 Physical Model Functions ......................................... 127
       7.6.7 Sequence Support Functions ..................................... 129

8 Nyquist Globals ................................................................. 130

9 Interactive Nyquist .............................................................. 134
   9.1 Interactive Control with the NyquistIDE .............................. 135
       9.1.1 Creating a Control Panel ......................................... 135
       9.1.2 Creating Controls .................................................. 135
       9.1.3 Accessing Control Values ........................................ 136
       9.1.4 Starting and Stopping Sounds .................................... 137
   9.2 Using Open Sound Control ............................................... 138
       9.2.1 Sending Open Sound Control Messages ....................... 139
       9.2.2 The ser-to-osc Program .......................................... 139

10 Time/Frequency Transformation ........................................... 140
Nyquist Reference Manual

11 MIDI, Adagio, and Sequences ................................................. 142
  11.1 The SEQ Type ............................................................... 142
  11.2 Adagio Score Language ................................................... 143
  11.3 Specifying Attributes ..................................................... 144
    11.3.1 Time ........................................................................ 144
    11.3.2 Pitch ....................................................................... 145
    11.3.3 Duration .................................................................... 145
    11.3.4 Next Time .................................................................. 146
    11.3.5 Rest .......................................................................... 146
    11.3.6 Articulation ............................................................... 146
    11.3.7 Loudness ................................................................. 147
    11.3.8 Voice ........................................................................ 147
    11.3.9 Timbre (MIDI Program) .............................................. 147
    11.3.10 Tempo ..................................................................... 148
    11.3.11 Rate ........................................................................ 148
  11.4 Default Attributes .............................................................. 149
  11.5 Examples .......................................................................... 149
  11.6 Advanced Features ............................................................ 152
    11.6.1 Time Units and Resolution ........................................... 152
    11.6.2 Multiple Notes Per Line .............................................. 152
    11.6.3 Control Change Commands ........................................ 153
    11.6.4 Multiple Tempi .......................................................... 154
    11.6.5 MIDI Synchronization ............................................... 155
    11.6.6 System Exclusive Messages ....................................... 155
    11.6.7 Control Ramps .......................................................... 156
    11.6.8 The !End Command ................................................... 157
    11.6.9 Calling C Routines ..................................................... 157
    11.6.10 Setting C Variables ................................................... 157
  12 Linear Prediction Analysis and Synthesis ................................. 159
    12.1 LPC Classes and Functions ............................................. 159
    12.2 Low-level LPC Functions .............................................. 160
  13 Developing and Debugging in Nyquist ....................................... 162
    13.1 Debugging ..................................................................... 162
    13.2 Useful Functions ............................................................ 163
  14 Xmusic and Algorithmic Composition ........................................ 166
    14.1 Xmusic Basics ............................................................... 166
    14.2 Xmusic Patterns ............................................................. 166
      14.2.1 Nested Patterns ...................................................... 167
      14.2.2 Periods ................................................................. 167
      14.2.3 General Parameters for Creating Pattern objects ............ 168
      14.2.4 cycle .................................................................... 168
      14.2.5 line ....................................................................... 169
      14.2.6 random ................................................................. 169

v 3.12
14.2.7 palindrome ......................................................... 169
14.2.8 heap .............................................................. 170
14.2.9 accumulation .................................................. 170
14.2.10 copier ........................................................... 171
14.2.11 accumulate ..................................................... 171
14.2.12 sum .............................................................. 172
14.2.13 product ......................................................... 172
14.2.14 eval ............................................................... 172
14.2.15 length ............................................................ 172
14.2.16 window .......................................................... 173
14.2.17 markov ........................................................... 173
14.3 Random Number Generators .................................. 174
14.4 Score Generation and Manipulation ............................ 180
  14.4.1 Keyword Parameters ........................................ 183
  14.4.2 Using score-gen ............................................... 184
  14.4.3 Score Manipulation .......................................... 185
  14.4.4 Xmusic and Standard MIDI Files ........................... 190
  14.4.5 Workspaces .................................................. 191
  14.4.6 Utility Functions .............................................. 192
15 Nyquist Libraries .................................................. 194
  15.1 Statistics ........................................................ 194
  15.2 Plots ............................................................... 194
  15.3 Labeling Audio Events, Marking Audio Times, Displaying Marked Audio Times 194
  15.4 Linear Regression .............................................. 195
  15.5 Vector Math, Linear Algebra .................................. 195
  15.6 Piano Synthesizer ................................................ 195
  15.7 Dynamics Compression ......................................... 195
  15.8 Clipping Softener ............................................... 197
  15.9 Graphical Equalizer ............................................ 197
  15.10 Sound Reversal ................................................ 197
  15.11 Time Delay Functions ......................................... 198
  15.12 Multiple Band Effects ......................................... 199
  15.13 Granular Synthesis ............................................ 199
  15.14 Chowning FM Voices ........................................... 200
  15.15 Atonal Melody Composition .................................. 200
  15.16 MIDI Utilities .................................................. 200
  15.17 Reverberation .................................................. 201
  15.18 DTMF Encoding ................................................. 201
  15.19 Dolby Surround(R), Stereo and Spatialization Effects ....... 201
  15.20 Drum Machine .................................................. 203
  15.21 Minimoog-inspired Synthesis ................................. 204
    15.21.1 Oscillator Parameters .................................. 204
    15.21.2 Noise Parameters ........................................ 205
Preface

This manual is a guide for users of Nyquist, a language for composition and sound synthesis. Nyquist grew out of a series of research projects, notably the languages Arctic and Canon. Along with Nyquist, these languages promote a functional style of programming and incorporate time into the language semantics.

Please help by noting any errors, omissions, or suggestions you may have. You can send your suggestions to Dannenberg@CS.CMU.EDU via email, or contact me by ordinary mail: Roger B. Dannenberg, School of Computer Science, Carnegie Mellon University, 5000 Forbes Ave., Pittsburgh, PA 15213-3890, USA.

Nyquist is a successor to Fugue, a language originally implemented by Chris Fraley, and extended by George Polly and Roger Dannenberg. Peter Velikonja and Dean Rubine were early users, and they proved the value as well as discovered some early problems of the system. This led to Nyquist, a reimplementation of Fugue by Roger Dannenberg with help from Joe Newcomer and Cliff Mercer. Ning Hu ported Zheng (Geoffrey) Hua and Jim Beauchamp’s piano synthesizer to Nyquist and also built NyqIDE, the Nyquist Integrated Development Environment for Windows. Dave Mowatt contributed the original version of NyquistIDE, the cross-platform interactive development environment. Dominic Mazzoni made a special version of Nyquist that runs within the Audacity audio editor, giving Nyquist a new interface and introducing Nyquist to many new users.

Many others have since contributed to Nyquist. Chris Tchou and Morgan Green worked on the Windows port. Eli Brandt contributed a number of filters and other synthesis functions. Pedro J. Morales, Eduardo Reck Miranda, Ann Lewis, and Erich Neuwirth have all contributed nyquist examples found in the demos folder of the Nyquist distribution. Philip Yam ported some synthesis functions from Perry Cook and Gary Scavone’s STK to Nyquist. Pedro Morales ported many more STK instruments to Nyquist. Dave Borel wrote the Dolby Pro-Logic encoding library and Adam Hartman wrote stereo and spatialization effects. Stephen Mangiat wrote the MiniMoog emulator. Phil Light recorded the drum samples and wrote drum machine software. The Xmusic library, particularly the pattern specification, was inspired by Rick Taube’s Common Music. The functions for generating probability distributions were implemented by Andreas Pfenning. John Chowning and Jorge Sastre contributed a SAL implementation of Chowning’s voice synthesis technique. Chen Ziheng implemented the majority of the cmuv phase vocoder library used for Nyquist’s phasevocoder and pv-pitch-time functions.

Starting with Version 3, Nyquist supports a version of SAL, providing an alternative to Lisp syntax. SAL was designed by Rick Taube, and the SAL implementation in Nyquist is based on Taube’s original implementation as part of his Common Music system.
The current NyquistIDE includes contributions from many. Chris Yealy and Derek D’Souza implemented early versions of the envelope editor. Daren Makuck and Michael Rivera wrote the original equalizer editor. Priyanka Raghavan implemented the sound browser. Dmitry Portnoy wrote the original "UPIC" editor.

Many others have made contributions, offered suggestions, and found bugs. If you were expecting to find your name here, I apologize for the omission, and please let me know.

I also wish to acknowledge support from CMU, Yamaha, and IBM for this work.
1 Introduction and Overview

Nyquist is a language for sound synthesis and music composition. Unlike score languages that tend to deal only with events, or signal processing languages that tend to deal only with signals and synthesis, Nyquist handles both in a single integrated system. Nyquist is also flexible and easy to use because it is based on an interactive Lisp interpreter.

With Nyquist, you can design instruments by combining functions (much as you would using the orchestra languages of Music V, cmusic, or Csound). You can call upon these instruments and generate a sound just by typing a simple expression. You can combine simple expressions into complex ones to create a whole composition.

Nyquist runs under Linux, Apple Mac OS X, Microsoft Windows NT, 2000, XP, and Vista, and it produces sound files or directly generates audio. Recent versions have also run on AIX, NeXT, SGI, DEC pmax, and Sun Sparc machines. (Makefiles for many of these are included, but out-of-date). Let me know if you have problems with any of these machines.

To use Nyquist, you should have a basic knowledge of Lisp. An excellent text by Touretzky is recommended [3]. Appendix C is the reference manual for XLISP, of which Nyquist is a superset. Starting with Version 3, Nyquist supports a variant of SAL. SAL is a language distinct from Lisp. It was introduced in Rick Taube’s Common Music system. Since there are some differences between Nyquist and Common Music, one should generally call this implementation "Nyquist SAL;" however, in this manual, I will just call it "SAL." SAL offers most of the capabilities of Lisp, but it uses an Algol-like syntax that may be more familiar to programmers with experience in Java, C, Basic, etc. An introduction to SAL is in Mary Simoni and Roger B. Dannenberg, Algorithmic Composition: A Guide to Composing Music with Nyquist [2].

1.1 Installation

Nyquist is a C++ program intended to run under various operating systems including Unix, Mac OS X, and Windows. Nyquist is based on Lisp, but it includes its own Lisp interpreter (a modified version of XLISP), so you do not need to install some other Lisp to run Nyquist. Other Lisp systems are not compatible with Nyquist.

Most Nyquist users run Nyquist under the Nyquist IDE, which is written in Java and depends on the Java runtime system. Most systems already have Java, but if you do not, you will need to install it. When you
install the Nyquist IDE, you will automatically get Nyquist and a set of runtime libraries.

There are generally two ways to install Nyquist:

- Get a pre-compiled version of the Nyquist IDE for Windows or Mac OS X. The Windows version comes packaged in an installer that installs and configures the Nyquist IDE. The Mac OS X version unpacks to a complete OS X application.

- Compile from sources. There is one set of sources for Mac, Windows, and Unix. Instructions for building applications from the sources are provided in the files `sys/win/README.txt`, `sys/mac/README.txt`, and `sys/unix/README.txt`.

You can download source code and precompiled versions from the Nyquist project on SourceForge (http://sourceforge.net/projects/nyquist). The latest source code can be obtained via Subversion (svn) using the following:

```
svn co svn://svn.code.sf.net/p/nyquist/code/trunk/nyquist nyquist
```

or by checking out nyquist using a graphical interface svn client such as TortoiseSVN for Windows.

### 1.1.1 Troubleshooting

All versions of Nyquist includes a Readme.txt file with additional information. Additional information and instructions on installation can be found in:

- `sys/win/README.txt`
- `sys/mac/README.txt`
- `sys/unix/README.txt`

### 1.2 Using NyquistIDE

The program named NyquistIDE is an "integrated development environment" for Nyquist. When you run NyquistIDE, it starts the Nyquist program and displays all Nyquist output in a window. NyquistIDE helps you by providing a Lisp and SAL editor, hints for command completion and function parameters, some graphical interfaces for editing envelopes and graphical equalizers, and a panel of buttons for common operations. A more complete description of NyquistIDE is in Chapter 2.

For now, all you really need to know is that you can enter Nyquist commands by typing into the upper left window. When you type return, the expression you typed is sent to Nyquist, and the results appear in the window below. You can edit files by clicking on the New File or Open File buttons. After editing some text, you can load the text into Nyquist by clicking the Load button. NyquistIDE always saves the file first; then it tells Nyquist to load the file. You will be prompted for a file name the first time you load a new file.
1.3 Using SAL

SAL mode means that Nyquist reads and evaluates SAL commands rather than Lisp. The SAL mode prompt is "SAL> " while the Lisp mode prompt is "> ". When Nyquist starts it normally enters SAL mode automatically, but certain errors may exit SAL mode. You can reenter SAL mode by typing the Lisp expression `sal`.

In SAL mode, you type commands in the SAL programming language. Nyquist reads the commands, compiles them into Lisp, and evaluates the commands. Commands can be entered manually by typing into the upper left text box in NyquistIDE.

1.4 Command Line

When run from the command line, Nyquist (usually the command is `ny`) looks for some options and some files to load. There are several options.

The `-R paths` option gives Nyquist a list of paths (separated by semicolons) to directories where Nyquist is allowed to read files. Files in any subdirectories of the listed directories can also be read, along with the directories themselves. This option can be used to restrict access to files.

The `-W paths` option gives Nyquist a list of paths (separated by semicolons) to directories where Nyquist is allowed to write files (including sound files). If `-W` is used, Nyquist will not open audio output to play sounds and will not allow access to Open Sound Control. This option can be used to protect files from accidentally malicious code. (If you are truly worried about Nyquist code, you should run Nyquist in a virtual machine or other isolated machine. It is unlikely that Nyquist is secure from all attacks.)

The `-L run-time-limit` option gives Nyquist a run time limit. If the limit is exceeded, Nyquist prints an error message and exits. The time is based on actual work rather than CPU seconds or wall time. One unit of run time is on the order of 10ms of CPU time (for a medium speed computer in 2014), but you can use the `GET-RUN-TIME` function to read the internal run-time counter to see how much computation you have done.

The `-M memory-limit` option gives Nyquist a memory limit. If the limit (in megabytes) is exceeded, Nyquist prints an error message and exits. The limit includes all XLISP data (cons cells, strings, etc.) plus sample blocks and tables allocated for signal processing.

The `-T transcript-file-name` option copies console output to a transcript file.

The `-V` option sets verbose mode. If set, any files specified on the command line to be loaded are loaded with the verbose option.

Additionally, the command line can name files to be loaded. Nyquist begins by loading `init.lsp`, which by default is in the `nyquist/runtime` directory. This file loads all the standard Nyquist functions. After that, Nyquist will load any other files named on the command line.

Finally, Nyquist expects to find search paths in the `XLISPPATH` environment variable (or for Windows, see installation instructions; the path is saved in the Windows Registry). Normally, you must set

v 3.12
### 1.5 Helpful Hints

Under Win95 and Win98, the console sometimes locks up. Activating another window and then reactivating the Nyquist window should unlock the output. (We suggest you use JNyqIDE, the interactive development environment rather than a console window.)

You can cut and paste text into Nyquist, but for serious work, you will want to use the Lisp `load` command. To save even more time, write a function to load your working file, e.g. `(defun l () (load "myfile.lsp"))`. Then you can type `(l)` to (re)load your file.

Using SAL, you can type

```lisp
define function l () load "myfile.lsp"
```

and then

```lisp
exec l()
```

to (re)load the file.

The Emacs editor is free GNU software and will help you balance parentheses if you use Lisp mode. In fact, you can run nyquist (without the IDE) as a subprocess to Emacs. A helpful discussion is at `http://www.audacity-forum.de/download/edgar/nyquist/nyquist-doc/examples/emacs/main.html`. If you use Emacs, there is also a SAL mode (the file is `sal-mode.el`) included with the Common Music distribution, which you can find on the Web at `sourceforge.net`.

The NyquistIDE also runs Nyquist as a subprocess and has built-in Lisp and SAL editors. If your editor does not help you balance parentheses, you may find yourself counting parens and searching for unbalanced expressions. If you are confused or desperate and using Lisp syntax, try the `:print t` option of the `load` function. By looking at the expressions printed, you should be able to tell where the last unbalanced expression starts. Alternatively, type `(file-sexprs)` and type the lisp file name at the prompt. This function will read and print expressions from the file, reporting an error when an extra paren or end-of-file is reached unexpectedly.

### 1.6 Using Lisp

Lisp mode means that Nyquist reads and evaluates Nyquist expressions in Lisp syntax. Since Nyquist is build on a Lisp interpreter, this is the "native" or machine language of Nyquist, and certain errors and functions may break out of the SAL interpreter, leaving you with a prompt for a Lisp expression. Alternatively, you can exit SAL simply by typing `exit` to get a Lisp prompt (`>`). Commands can be entered manually by typing into the upper left text box in NyquistIDE.
1.7 Examples

We will begin with some simple Nyquist programs. Throughout the manual, we will assume SAL mode and give examples in SAL, but it should be emphasized that all of these examples can be performed using Lisp syntax. See Section 6.2 on the relationship between SAL and Lisp.

Detailed explanations of the functions used in these examples will be presented in later chapters, so at this point, you should just read these examples to get a sense of how Nyquist is used and what it can do. The details will come later. Most of these examples can be found in the file nyquist/demos/examples.sal. Corresponding Lisp syntax examples are in the file nyquist/demos/examples.lsp.

Our first example makes and plays a sound:

```
;; Making a sound.
play osc(60) ; generate a loud sine wave
```

This example is about the simplest way to create a sound with Nyquist. The `osc` function generates a sound using a table-lookup oscillator. There are a number of optional parameters, but the default is to compute a sinusoid with an amplitude of 1.0. The parameter 60 designates a pitch of middle C. (Pitch specification will be described in greater detail later.) The result of the `osc` function is a sound. To hear a sound, you must use the `play` command, which plays the file through the machine’s D/A converters. It also writes a soundfile in case the computation cannot keep up with real time. You can then (re)play the file by typing:

```
exec r()
```

This `r()` function is a general way to "replay" the last thing written by `play`.

Note: when Nyquist plays a sound, it scales the signal by $2^{15} - 1$ and (by default) converts to a 16-bit integer format. A signal like `(osc 60)`, which ranges from +1 to -1, will play as a full-scale 16-bit audio signal.

1.7.1 Waveforms

Our next example will be presented in several steps. The goal is to create a sound using a wavetable consisting of several harmonics as opposed to a simple sinusoid. In order to build a table, we will use a function that computes a single harmonic and add harmonics to form a wavetable. An oscillator will be used to compute the harmonics.

The function `mkwave` calls upon `build-harmonic` to generate a total of four harmonics with amplitudes 0.5, 0.25, 0.125, and 0.0625. These are scaled and added (using `+`) to create a waveform which is bound temporarily to `*table*`.

A complete Nyquist waveform is a list consisting of a sound, a pitch, and T, indicating a periodic waveform. The pitch gives the nominal pitch of the sound. (This is implicit in a single cycle wave table, but a sampled sound may have many periods of the fundamental.) Pitch is expressed in half-steps, where middle C is 60 steps, as in MIDI pitch numbers. The list of sound, pitch, and T is formed in the last line of `mkwave`: since `build-harmonic` computes signals with a duration of one second, the fundamental is 1 Hz, and the `hz-to-step` function converts to pitch (in units of steps) as required.
define function mkwave()
    begin
        set *table* = 0.5 * build-harmonic(1, 2048) +
            0.25 * build-harmonic(2, 2048) +
            0.125 * build-harmonic(3, 2048) +
            0.0625 * build-harmonic(4, 2048)
        set *table* = list(*table*, hz-to-step(1.0), #t)
    end

Now that we have defined a function, the last step of this example is to build the wave. The following code calls mkwave the first time the code is executed (loaded from a file). The second time, the variable *mkwave* will be true, so mkwave will not be invoked:

    if ! fboundp(quote(*mkwave*)) then
        begin
            exec mkwave()
            set *mkwave* = #t
        end

1.7.2 Wavetables

When Nyquist starts, several waveforms are created and stored in global variables for convenience. They are: *sine-table*, *saw-table*, and *tri-table*, implementing sinusoid, sawtooth, and triangle waves, respectively. The variable *table* is initialized to *sine-table*, and it is *table* that forms the default wave table for many Nyquist oscillator behaviors. If you want a proper, band-limited waveform, you should construct it yourself, but if you do not understand this sentence and/or you do not mind a bit of aliasing, give *saw-table* and *tri-table* a try.

Note that in Lisp and SAL, global variables often start and end with asterisks (*). These are not special syntax, they just happen to be legal characters for names, and their use is purely a convention.

1.7.3 Sequences

Finally, we define my-note to use the waveform, and play several notes in a simple score. Note that the function my-note has only one command (a return command), so it is not necessary to use begin and end. These are only necessary when the function body consists of a sequence of statements:

    define function my-note(pitch, dur)
        return osc(pitch, dur, *table*)
    end

    play seq(my-note(c4, i), my-note(d4, i), my-note(f4, i),
        my-note(g4, i), my-note(d4, q))

Here, my-note is defined to take pitch and duration as parameters; it calls osc to do the work of generating a waveform, using *table* as a wave table.
The `seq` function is used to invoke a sequence of behaviors. Each note is started at the time the previous note finishes. The parameters to `my-note` are predefined in Nyquist: `c4` is middle C, `i` (for eighth note) is 0.5, and `q` (for Quarter note) is 1.0. See Section 1.8 for a complete description. The result is the sum of all the computed sounds.

Sequences can also be constructed using the `at` transformation to specify time offsets. See demos/sequence_example.htm demos, sequence for more examples and explanation.

### 1.7.4 Envelopes

The next example will illustrate the use of envelopes. In Nyquist, envelopes are just ordinary sounds (although they normally have a low sample rate). An envelope is applied to another sound by multiplication using the `mult` function. The code shows the definition of `env-note`, defined in terms of the `note` function in the previous example. In `env-note`, a 4-phase envelope is generated using the `env` function, which is illustrated in Figure 1.1.

```plaintext
; env-note produces an enveloped note. The duration
; defaults to 1.0, but stretch can be used to change
; the duration.
; Uses my-note, defined above.
;
define function env-note(p)  
  return my-note(p, 1.0) *  
    env(0.05, 0.1, 0.5, 1.0, 0.5, 0.4)

; try it out:
;
play env-note(c4)
```

Figure 1.1: An envelope generated by the `env` function.
While this example shows a smooth envelope multiplied by an audio signal, you can also multiply audio signals to achieve what is often called **ring modulation**. See the code and description in `demos/scratch_tutorial.htm` for an interesting use of ring modulation to create "scratch" sounds.

In the next example, The `stretch` operator (`~`) is used to modify durations:

```lisp
; now use stretch to play different durations
play seq(seq(env-note(c4), env-note(d4)) ~ 0.25,
         seq(env-note(f4), env-note(g4)) ~ 0.5,
         env-note(c4))
```

In addition to `stretch`, there are a number of transformations supported by Nyquist, and transformations of abstract behaviors is perhaps the fundamental idea behind Nyquist. Chapter 3 is devoted to explaining this concept, and further elaboration can be found elsewhere [1].

### 1.7.5 Piece-wise Linear Functions

It is often convenient to construct signals in Nyquist using a list of (time, value) breakpoints which are linearly interpolated to form a smooth signal. Envelopes created by `env` are a special case of the more general piece-wise linear functions created by `pwl`. Since `pwl` is used in some examples later on, we will take a look at `pwl` now. The `pwl` function takes a list of parameters which denote (time, value) pairs. There is an implicit initial (time, value) pair of (0, 0), and an implicit final value of 0. There should always be an odd number of parameters, since the final value (but not the final time) is implicit. Here are some examples:

```lisp
; symmetric rise to 10 (at time 1) and fall back to 0 (at time 2):
; ...
pwl(1, 10, 2)

; a square pulse of height 10 and duration 5.
; Note that the first pair (0, 10) overrides the default initial
; point of (0, 0). Also, there are two points specified at time 5:
; (5, 10) and (5, 0). (The last 0 is implicit). The conflict is
; automatically resolved by pushing the (5, 10) breakpoint back to
; the previous sample, so the actual time will be 5 - 1/sr, where
; sr is the sample rate.
; ...
pwl(0, 10, 5, 10, 5)

; a constant function with the value zero over the time interval
; 0 to 3.5. This is a very degenerate form of pwl. Recall that there
; is an implicit initial point at (0, 0) and a final implicit value of
; 0, so this is really specifying two breakpoints: (0, 0) and (3.5, 0):
; ...
pwl(3.5)
Nyquist Reference Manual

Chapter 1. Introduction and Overview

; a linear ramp from 0 to 10 and duration 1.
; Note the ramp returns to zero at time 1. As with the square pulse
; above, the breakpoint (1, 10) is pushed back to the previous sample.
; pwl(1, 10, 1)

; If you really want a linear ramp to reach its final value at the
; specified time, you need to make a signal that is one sample longer.
; The RAMP function does this:
; ramp(10); ramp from 0 to 10 with duration 1 + one sample period

; RAMP is based on PWL; it is defined in nyquist.lsp.

1.8 Predefined Constants

For convenience and readability, Nyquist pre-defines some constants, mostly based on the notation of the
Adagio score language, as follows:

- **Dynamics** Note: these dynamics values are subject to change.

  lppp = -12.0 (dB)
  lpp = -9.0
  lp = -6.0
  lmp = -3.0
  lmf = 3.0
  lf = 6.0
  lff = 9.0
  lfff = 12.0
  dB0 = 1.00
  dB1 = 1.122
  dB10 = 3.1623

- **Durations**

  s = Sixteenth = 0.25
  i = Eighth = 0.5
  q = Quarter = 1.0
  h = Half = 2.0
  w = Whole = 4.0
  sd, id, qd, hd, wd = dotted durations.
• **Pitches** Pitches are based on an A4 of 440Hz. To achieve a different tuning, set *A4-Hertz* to the desired frequency for A4, and call (set-pitch-names). This will recompute the names listed below with a different tuning. In all cases, the pitch value 69.0 corresponds exactly to 440Hz, but fractional values are allowed, so for example, if you set *A4-Hertz* to 444 (Hz), then the symbol A4 will be bound to 69.1567, and C4 (middle C), which is normally 60.0, will be 60.1567.

\[
\begin{align*}
c0 &= 12.0  
cs0, \ df0 &= 13.0  
d0 &= 14.0  
ds0, \ ef0 &= 15.0  
e0 &= 16.0  
f0 &= 17.0  
fs0, \ gf0 &= 18.0  
g0 &= 19.0  
gs0, \ af0 &= 20.0  
a0 &= 21.0  
as0, \ bf0 &= 22.0  
b0 &= 23.0  
c1 \ldots \ b1 &= 24.0 \ldots 35.0  
c2 \ldots \ b2 &= 36.0 \ldots 47.0  
c3 \ldots \ b3 &= 48.0 \ldots 59.0  
c4 \ldots \ b4 &= 60.0 \ldots 71.0  
c5 \ldots \ b5 &= 72.0 \ldots 83.0  
c6 \ldots \ b6 &= 84.0 \ldots 95.0  
c7 \ldots \ b7 &= 96.0 \ldots 107.0  
c8 \ldots \ b8 &= 108.0 \ldots 119.0
\end{align*}
\]

• **Miscellaneous**

\[ny:\text{all} = "all\ the\ samples" \text{ (i.e. a big number)} = 1000000000\]

### 1.9 More Examples

More examples can be found in the directory demos, part of the standard Nyquist release. In the Apple OS X version of Nyquist, the demos and doc directories are inside the NyquistIDE application bundle. To make it easier to access these hidden files, the NyquistIDE installs links to these directories the first time you run it. The links will be in the same directory as the NyquistIDE itself.
The file demos/index.htm is an index to all the demo descriptions. In this directory, you will find the following and more:

- Code to create atonal melodies (demos/src/atonal-melodies.sal).
- How to make arpeggios (demos/arpeggiator.htm and demos/src/arp.sal)
- Gong sounds by additive synthesis (demos/pmorales/b1.lsp and demos/mateos/gong.lsp)
- Risset's spectral analysis of a chord (demos/pmorales/b2.lsp)
- Bell sounds (demos/pmorales/b3.lsp, demos/pmorales/e2.lsp, demos/pmorales/partial.lsp, and demos/mateos/bell.lsp)
- Drum sounds by Risset (demos/pmorales/b8.lsp)
- Shepard tones (demos/src/shepard.lsp and demos/pmorales/b9.lsp)
- Random signals (demos/pmorales/c1.lsp)
- Buzz with formant filters (demos/pmorales/buzz.lsp)
- Computing samples directly in Lisp (using Karplus-Strong and physical modelling as examples) (demos/pmorales/d1.lsp)
- FM Synthesis examples, including FM voices (demos/src/fm-voices-chowning.sal) designed by John Chowning, bell, wood drum, brass sounds, tuba sound (demos/mateos/tuba.lsp) and clarinet sounds (demos/pmorales/e2.lsp)
- Rhythmic patterns (demos/rhythm_tutorial.htm)
- Drum Samples and Drum Machine (demos/plight/drum.lsp). (See Section 15.20).
2 The NyquistIDE Program

The NyquistIDE program combines many helpful functions and interfaces to help you get the most out of Nyquist. NyquistIDE is implemented in Java, and you will need the Java runtime system or development system installed on your computer to use NyquistIDE. The best way to learn about NyquistIDE is to just use it. This chapter introduces some of the less obvious features. If you are confused by something and you do not find the information you need here, please contact the author.

2.1 NyquistIDE Overview

The NyquistIDE runs the command-line version of Nyquist as a subtask, so everything that works in Nyquist should work when using the NyquistIDE and vice-versa. Input to Nyquist is usually entered in the top left window of the NyquistIDE. When you type return, if the expression or statement appears to be complete, the expression you typed is sent to Nyquist. Output from Nyquist appears in a window below. You cannot type into or edit the output window text.

The normal way to use the NyquistIDE is to create or open one or more files. You edit these files and then click the Load button. To load a file, NyquistIDE saves the file, sets the current directory of Nyquist to that of the file, then issues a load command to Nyquist. In this case and several others, you may notice that NyquistIDE sends expressions to Nyquist automatically for evaluation. You can always see the commands and their results in the output window.

Notice that when you load a selected file window, NyquistIDE uses setdir to change Nyquist’s current directory. This helps to keep the two programs in sync. Normally, you should keep all the files of a project in the same directory and avoid manually changing Nyquist’s current directory (i.e. avoid calling setdir in your code).

Arranging windows in the NyquistIDE can be time-consuming, and depending on the operating system, it is possible for a window to get into a position where you cannot drag it to a new position. The Window:Tile menu command can be used to automatically lay out windows in a rational way. There is a preference setting to determine the height of the completion list relative to the height of the output window.

Many of the key commands for editing files are standard and built into Java. In particular, Control-Z is the "undo" command, even on OS X where you would expect to use Command-Z. A special command, "Copy to
Lisp," or Control-U (Command-U on OS x) copies the selection and enters it into the input window as if it had been typed there.

## 2.2 The Button Bar

There are a number of buttons with frequently-used operations. These are:

- **Info** – Print information about Nyquist memory utilization, including the number of free cons cells, the number of garbage collections, the total number of cons cells, the total amount of sample buffer memory, and the amount of memory in free sample buffers.

- **Break** – Send a break character to XLISP. This can be used to enter the debugger (the break loop) while a program is running. Resume by typing `(co).`

- **SAL/Lisp** – Switch modes. The button names the mode (SAL or Lisp) you will switch to, not the current mode. For example, if you are in Lisp mode and want to type a SAL command, click the SAL button first.

- **Top** – Enters `(top)` into Nyquist. If the XLISP prompt is 1> or some other integer followed by ">", clicking the Top button will exit the debug loop and return to the top-level prompt.

- **Replay** – Enters `(r)` into Nyquist. This command replays the last computed sound.

- **F2-F12** – Enters `(f2)` etc. into Nyquist. These commands are not built-in, and allow users to define their own custom actions.

- **Browse** – Equivalent to the Window:Browse menu item. (See Section 2.4.)

- **EQ** – Equivalent to the Window:EQ menu item. (See Section 2.6.)

- **EnvEdit** – Equivalent to the Window:Envelope Edit menu item. (See Section 2.5.)

- **NewFile** – Equivalent to the File:New menu item. Opens a new file editing window for creating and loading a Lisp or SAL program file.

- **OpenFile** – Equivalent to the File:Open menu item. Opens an existing Lisp or SAL program file for editing and loading.

- **SaveFile** – Equivalent to the File:Save menu item (found on the editing window’s menu bar). Saves the contents of an editing window to its associated file.

- **Load** – Equivalent to the File:Load menu item (found on the editing window’s menu bar). Performs a Save operation, then sends a command to Nyquist that loads the file as a program.

- **Mark** – Sends a Control-A to Nyquist. While playing a sound, this displays and records the approximate time in the audio stream. (See Section 7.5 for more detail.)
2.3 Command Completion

To help with programming, NyquistIDE maintains a command-completion window. As you type the first letters of function names, NyquistIDE lists matching functions and their parameters in the Completion List window. If you click on an entry in this window, the displayed expression will replace the incompletely typed function name. A preference allows you to match initial letters or any substring of the complete function name. This is controlled by the "Use full search for code completion" preference.

In addition, if you right click (or under Mac OS X, hold down the Alt/Option key and click) on an entry, NyquistIDE will display documentation for the function. Documentation can come from a local copy or from the online copy (determined by the "Use online manual instead of local copy" preference). Documentation can be displayed within the NyquistIDE window or in an external browser (determined by the "Use window in NyquistIDE for help browser" preference.) Currently, the external browser option does not seem to locate documentation properly, but this should be fixed in the future.

2.4 Browser

If you click on the Browse button or use the Window:Browse menu command, NyquistIDE will display a browser window that is pre-loaded with a number of Nyquist commands to create sounds. You can adjust parameters, audition the sounds, and capture the expression that creates the sound. In many cases, the expression checks to see if necessary functions are defined, loading files if necessary before playing the sound. If you want to use a sound in your own program, you can often simplify things by explicitly loading the required file just once at the beginning of your file.

Since Nyquist now supports a mix of Lisp and SAL, you may find yourself in the position of having code from the browser in one language while you are working in the other. The best way to handle this is to put the code for the sound you want into a function defined in a Lisp (.lsp) or SAL (.sal) file. Load the file (from Lisp, use the sal-load command to load a SAL file), and call the function from the language of your choice.

2.5 Envelope Editor

The envelope editor allows you graphically to design and edit piece-wise linear and exponential envelopes. The editor maintains a list of envelopes and you select the one to edit or delete using the drop down list in the Saved Envelopes List area. The current envelope appears in the Graphical Envelope Editor area. You can click to add or drag points. Alternatively, you can use the Envelope Points window to select and edit any breakpoint by typing coordinates. The duration of the envelope is controlled by the Stop field in the Range area, and the vertical axis is controlled by the Min and Max fields.

When you click the Save button, all envelopes are written to Nyquist. You can then use the envelope by treating the envelope name as a function. For example, if you define an envelope named "fast-attack," then you can create the envelope within a Nyquist SAL program by writing the expression fast-attack().
These edited envelopes are saved to a file named `workspace.lsp` in the current directory. The workspace is Nyquist’s mechanism for saving data of all kinds (see Section 14.4.5). The normal way to work with workspaces is to (1) load the workspace, i.e. `load "workspace"`, as soon as you start Nyquist; (2) invoke the envelope editor to change values in the workspace; and (3) save the workspace at any time, especially before you exit NyquistIDE. If you follow these steps, envelopes will be preserved from session to session, and the entire collection of envelopes will appear in the editor. Be sure to make backups of your `workspace.lsp` file along with your other project files.

The envelope editor can create linear and exponential envelopes. Use the Type pull-down menu to select the type you want. Envelopes can be created using default starting and ending values using `pwl` or `pwe`, or you can specify the initial values using `pwlv` or `pwev`. The envelope editor uses `pwl` or `pwe` if no point is explicitly entered as the initial or final point. To create a `pwlv` or `pwev` function, create a point and drag it to the leftmost or rightmost edge of the graphical editing window. You will see the automatically generated default starting or ending point disappear from the graph.

Exponential envelopes should never decay to zero. If you enter a zero amplitude, you will see that the envelope remains at zero to the next breakpoint. To get an exponential decay to "silence," try using an amplitude of about 0.001 (about -60dB). To enter small values like this, you can type them into the Amplitude box and click "Update Point."

The Load button refreshes the editor from data saved in the Nyquist process. Normally, there is no need to use this because the editor automatically loads data when you open it.

### 2.6 Equalizer Editor

The Equalizer Editor provides a graphical EQ interface for creating and adjusting equalizers. Unlike the envelope editor, where you can type any envelope name, equalizers are named `eq-0`, `eq-1`, etc., and you select the equalizer to edit using a pull-down menu. The Set button should be use to record changes.

### 2.7 UPIC Editor

The UPIC Editor is inspired by the UPIC system by Iannis Xenakis at the Centre d’Edudes de Mathematique et Automatique Musicales (CEMaMu). The UPIC Editor is accessed by the "Upic Edit" menu item in the "Window" menu of the NyquistIDE. Once opened, you can draw pitch contours in the main panel by pressing the left mouse button and dragging with the mouse down. Contours represent tones in a frequency vs. time coordinate system. Any contour can be deleted by right-clicking (or shift-clicking on an Apple computer) to select the contour (indicated by the color red), and typing the Delete key.

A collection of contours can be saved to a file and later retrieved using the items in the File menu (use the File menu in the UPIC Editor window, not in the main NyquistIDE window.) The file is a SAL program in a special format that can be parsed by the UPIC Editor. The file can also be loaded into Nyquist using the File:Load menu item, or by executing a load command in Nyquist.
The panel at the top of the editor offers control over various parameters. The Name box is a Nyquist variable name. This name takes effect when you save a file from the UPIC Editor. The variable name is stored in the file so that when a UPIC Editor-generated file is loaded into Nyquist, the data is assigned to this variable name. The data is a list of contours, where each contour specifies a waveform, an envelope, and a list of time-frequency coordinates.

The next item in the panel is the Waveform box. The Waveform box names a waveform for a contour. Default waveforms are sinusoid, triangle, and sawtooth, but you can type in your own names. The currently selected waveform is stored with the contour when it is created (entered by drawing). You cannot change or edit the waveform name associated with a contour once the contour is created, but you can always delete the contour and replace it. The Envelope box names an envelope for a contour. The envelope names a Nyquist function. The default, upic-env is a trapezoid shape with an onset time and offset time of 10ms. As with waveforms, the envelope is stored with each contour when the contour is created and cannot be edited.

The Stop Time box gives the duration of the drawing area in seconds. The Min Freq box gives the minimum frequency (at the bottom of the drawing area), and the Max Freq box gives the maximum frequency (at the top of the drawing area). The vertical frequency axis can use a linear scale corresponding to frequency in Hertz or a logarithmic scale corresponding to semitones. The "linear" checkbox selects the linear scale. When any of these parameters (described in this paragraph and delimited by the border labeled "Range" on the control panel) is changed, you must press the Update Range button for the change to take effect.

The Background menu lets you display a grid that indicates pitch locations. The "C's" item draws a line at C in every visible octave. E.g. middle C is about 260 Hz, so a reference line will be drawn near 260 Hz. Lines will be drawn around 130 Hz (an octave below middle C), and around 520 Hz (an octave above middle C), etc. The "GrandStaff" menu item draws reference lines for each line of the grand staff commonly used for piano music. The pitches are G2, B2, D3, F3, A3, E4, G4, B4, D5, and F5. Finally, you can load a picture using the Background:Load Picture... menu item. Then, the Background:Show Picture menu item toggles whether the picture is displayed or not. This feature allows you to trace an image. (For example, see the Sonic Self-Portrait at http://www.cs.cmu.edu/~rbd.) You may wish to use an image editor to lighten the image so that superimposed contours will be more visible.

Each change to the Range data, background choice, and each entry of a contour is an action that you can undo or redo with the Undo and Redo buttons.

To convert UPIC data into sound, first load upic.sal and load a file generated by the UPIC Editor. Now, suppose the variable name used is upicdata. You can play the data by writing

\[
\text{play upic(upicdata)}
\]

If you created your own names for waveforms or envelopes, you must be sure that these exist before calling the upic function. Each waveform must be the name of a variable which is set to a Nyquist wave table. (See Section 1.7.1 for information on how to create a wave table.) Also, each envelope must name a function with no parameters that will return an amplitude envelope. The following is the built-in definition for upic-env:

\[
\text{define function upic-env()}
\]

\[
\text{return env(0.01, 0.01, 0.01, 1, 1, 1)}
\]

To make a custom envelope function named upic-smooth with a 0.2 second attack and a 0.3 second decay, you could write:

\[
\text{v 3.12}
\]
define function upic-smooth()
    return env(0.2, 0.01, 0.3, 1, 1, 1)
3 Behavioral Abstraction

In Nyquist, all functions are subject to transformations. You can think of transformations as additional parameters to every function, and functions are free to use these additional parameters in any way. The set of transformation parameters is captured in what is referred to as the *transformation environment*. (Note that the term *environment* is heavily overloaded in computer science. This is yet another usage of the term.)

Behavioral abstraction is the ability of functions to adapt their behavior to the transformation environment. This environment may contain certain abstract notions, such as loudness, stretching a sound in time, etc. These notions will mean different things to different functions. For example, an oscillator should produce more periods of oscillation in order to stretch its output. An envelope, on the other hand, might only change the duration of the sustain portion of the envelope in order to stretch. Stretching a sample could mean resampling it to change its duration by the appropriate amount.

Thus, transformations in Nyquist are not simply operations on signals. For example, if I want to stretch a note, it does not make sense to compute the note first and then stretch the signal. Doing so would cause a drop in the pitch. Instead, a transformation modifies the *transformation environment* in which the note is computed. Think of transformations as making requests to functions. It is up to the function to carry out the request. Since the function is always in complete control, it is possible to perform transformations with "intelligence;" that is, the function can perform an appropriate transformation, such as maintaining the desired pitch and stretching only the "sustain" portion of an envelope to obtain a longer note.

3.1 The Environment

The transformation environment consists of a set of special variables. These variables should not be read directly and should *never* be set directly by the programmer. Instead, there are functions to read them, and they are automatically set and restored by transformation operators, which will be described below.

The transformation environment consists of the following elements. Although each element has a "standard interpretation," the designer of an instrument or the composer of a complex behavior is free to interpret the environment in any way. For example, a change in *loud* may change timbre more than amplitude, and *transpose* may be ignored by percussion instruments:
*warp*
Time transformation, including time shift, time stretch, and continuous time warp. The value of *warp* is interpreted as a function from logical (local score) time to physical (global real) time. Do not access *warp* directly. Instead, use local-to-global(t) to convert from a logical (local) time to real (global) time. Most often, you will call local-to-global(0). Several transformation operators operate on *warp*, including at (@), stretch (\~), and warp. See also get-duration() and get-warp().

*loud*
Loudness, expressed in decibels. The default (nominal) loudness is 0.0 dB (no change). Do not access *loud* directly. Instead, use get-loud() to get the current value of *loud* and either loud or loud-abs to modify it.

*transpose*
Pitch transposition, expressed in semitones. (Default: 0.0). Do not access *transpose* directly. Instead, use get-transpose() to get the current value of *transpose* and either transpose or transpose-abs to modify it.

*sustain*
The "sustain," "articulation," "duty factor," or amount by which to separate or overlap sequential notes. For example, staccato might be expressed with a *sustain* of 0.5, while very legato playing might be expressed with a *sustain* of 1.2. Specifically, *sustain* stretches the duration of notes (sustain) without affecting the inter-onset time (the rhythm). Do not access *sustain* directly. Instead, use get-sustain() to get the current value of *sustain* and either sustain or sustain-abs to modify it.

*start*
Start time of a clipping region. Note: unlike the previous elements of the environment, *start* has a precise interpretation: no sound should be generated before *start*. This is implemented in all the low-level sound functions, so it can generally be ignored. You can read *start* directly, but use extract or extract-abs to modify it. Note 2: Due to some internal confusion between the specified starting time and the actual starting time of a signal after clipping, *start* is not fully implemented.

*stop*
Stop time of clipping region. By analogy to *start*, no sound should be generated after this time. *start* and *stop* allow a composer to preview a small section of a work without computing it from beginning to end. You can read *stop* directly, but use extract or extract-abs to modify it. Note: Due to some internal confusion between the specified starting time and the actual starting time of a signal after clipping, *stop* is not fully implemented.

*control-srate*
Sample rate of control signals. This environment element provides the default sample rate for control signals. There is no formal distinction between a control signal and an audio signal. You can read *control-srate* directly, but use control-srate or control-srate-abs to modify it.
3.2 Sequential Behavior

Previous examples have shown the use of `seq`, the sequential behavior operator. We can now explain `seq` in terms of transformations. Consider the simple expression:

```lisp
play seq(my-note(c4, q), my-note(d4, i))
```

The idea is to create the first note at time 0, and to start the next note when the first one finishes. This is all accomplished by manipulating the environment. In particular, `*warp*` is modified so that what is locally time 0 for the second note is transformed, or warped, to the logical stop time of the first note.

One way to understand this in detail is to imagine how it might be executed: first, `*warp*` is set to an initial value that has no effect on time, and `my-note(c4, q)` is evaluated. A sound is returned and saved. The sound has an ending time, which in this case will be 1.0 because the duration `q` is 1.0. This ending time, 1.0, is used to construct a new `*warp*` that has the effect of shifting time by 1.0. The second note is evaluated, and will start at time 1. The sound that is returned is now added to the first sound to form a composite sound, whose duration will be 2.0. `*warp*` is restored to its initial value.

Notice that the semantics of `seq` can be expressed in terms of transformations. To generalize, the operational rule for `seq` is: evaluate the first behavior according to the current `*warp*`. Evaluate each successive behavior with `*warp*` modified to shift the new note’s starting time to the ending time of the previous behavior. Restore `*warp*` to its original value and return a sound which is the sum of the results.

In the Nyquist implementation, audio samples are only computed when they are needed, and the second part of the `seq` is not evaluated until the ending time (called the logical stop time) of the first part. It is still the case that when the second part is evaluated, it will see `*warp*` bound to the ending time of the first part.

A language detail: Even though Nyquist defers evaluation of the second part of the `seq`, the expression can reference variables according to ordinary Lisp/SAL scope rules. This is because the `seq` captures the expression in a closure, which retains all of the variable bindings.

3.3 Simultaneous Behavior

Another operator is `sim`, which invokes multiple behaviors at the same time. For example,

```lisp
play 0.5 * sim(my-note(c4, q), my-note(d4, i))
```
will play both notes starting at the same time.

The operational rule for \texttt{sim} is: evaluate each behavior at the current *warp* and return the sum of the results. (In SAL, the \texttt{sim} function applied to sounds is equivalent to adding them with the infix + operator. The following section illustrates two concepts: first, a sound is not a behavior, and second, the \texttt{sim} operator and the \texttt{at} transformation can be used to place sounds in time.

### 3.4 Sounds vs. Behaviors

The following example loads a sound from a file in the current directory and stores it in \texttt{a-snd}:

```plaintext
; load a sound
; set a-snd = s-read(strcat(current-path(), "demo-snd.aiff"))
```

; play it
; play a-snd

One might then be tempted to write the following:

```plaintext
play seq(a-snd, a-snd) ;WRONG!
```

Why is this wrong? Recall that \texttt{seq} works by modifying *warp*, not by operating on sounds. So, \texttt{seq} will proceed by evaluating \texttt{a-snd} with different values of *warp*. However, the result of evaluating \texttt{a-snd} (a variable) is always the same sound, regardless of the environment; in this case, the second \texttt{a-snd} should start at time 0.0, just like the first. In this case, after the first sound ends, Nyquist is unable to "back up" to time zero, so in fact, this will play two sounds in sequence, but that is a result of an implementation detail rather than correct program execution. In fact, a future version of Nyquist might (correctly) stop and report an error when it detects that the second sound in the sequence has a real start time that is before the requested one.

How then do we obtain a sequence of two sounds properly? What we really need here is a behavior that transforms a given sound according to the current transformation environment. That job is performed by \texttt{cue}. For example, the following will behave as expected, producing a sequence of two sounds:

```plaintext
play seq(cue(a-snd), cue(a-snd))
```

This example is correct because the second expression will shift the sound stored in \texttt{a-snd} to start at the end time of the first expression.

The lesson here is very important: **sounds are not behaviors!** Behaviors are computations that generate sounds according to the transformation environment. Once a sound has been generated, it can be stored, copied, added to other sounds, and used in many other operations, but sounds are not subject to transformations. To transform a sound, use \texttt{cue}, \texttt{sound}, or \texttt{control}. The differences between these operations are discussed later. For now, here is a "cue sheet" style score that plays 4 copies of \texttt{a-snd}:
Nyquist Reference Manual

3.5 The At Transformation

The second concept introduced by the previous example is the @ operation, which shifts the *warp* component of the environment. For example,

```
cue(a-snd) @ 0.7
```

can be explained operationally as follows: modify *warp* by shifting it by 0.7 and evaluate cue(a-snd). Return the resulting sound after restoring *warp* to its original value. Notice how @ is used inside a sim construct to locate copies of a-snd in time. This is the standard way to represent a note-list or a cue-sheet in Nyquist.

This also explains why sounds need to be cue’d in order to be shifted in time or arranged in sequence. If this were not the case, then sim would take all of its parameters (a set of sounds) and line them up to start at the same time. But cue(a-snd) @ 0.7 is just a sound, so sim would "undo" the effect of @, making all of the sounds in the previous example start simultaneously, in spite of the @! Since sim respects the intrinsic starting times of sounds, a special operation, cue, is needed to create a new sound with a new starting time.

3.6 The Stretch Transformation

In addition to At (denoted in SAL by the @ operator, the Stretch transformation is very important. It appeared in the introduction, and it is denoted in SAL by the ~ operator (or in LISP by the stretch special form). Stretch also operates on the *warp* component of the environment. For example,

```
osc(c4) ~ 3
```
do the following: modify *warp*, scaling the degree of "stretch" by 3, and evaluate osc(c4). The osc behavior uses the stretch factor to determine the duration, so it will return a sound that is 3 seconds long. Restore *warp* to its original value. Like At, Stretch only affects behaviors. a-snd ~ 10 is equivalent to a-snd because a-snd is a sound, not a behavior. Behaviors are functions that compute sounds according to the environment and return a sound.
3.7 Nested Transformations

Transformations can be combined using nested expressions. For example,

\[
sim(cue(a\text{-snd}),
    \quad \text{loud}(6.0, \text{cue}(a\text{-snd}) @ 3))
\]

scales the amplitude as well as shifts the second entrance of \text{a\text{-snd}}.

Why use \text{loud} instead of simply multiplying \text{a\text{-snd}} by some scale factor? Using \text{loud} gives the behavior the chance to implement the abstract property \textit{loudness} in an appropriate way, e.g. by including timbral changes. In this case, the behavior is \text{cue}, which implements \textit{loudness} by simple amplitude scaling, so the result is equivalent to multiplication by \text{db-to-linear}(6.0).

Transformations can also be applied to groups of behaviors:

\[
\text{loud}(6.0, \text{sim}(\text{cue}(a\text{-snd}) @ 0.0, \text{cue}(a\text{-snd}) @ 0.7))
\]

3.8 Defining Behaviors

Groups of behaviors can be named using \text{define} (we already saw this in the definitions of \text{my\text{-note} and env\text{-note}}). Here is another example of a behavior definition and its use. The definition has one parameter:

\[
\text{define function snds(dly)}
\]

\[
\begin{align*}
\text{return } & \sim(\text{cue}(a\text{-snd}) @ 0.0, \\
& \text{cue}(a\text{-snd}) @ 0.7, \\
& \text{cue}(a\text{-snd}) @ 1.0, \\
& \text{cue}(a\text{-snd}) @ (1.2 + dly))
\end{align*}
\]

\text{play snds}(0.1)
\text{play loud}(0.25, \text{snds}(0.3) \sim 0.9)

In the last line, \text{snds} is transformed: the transformations will apply to the \text{cue} behaviors within \text{snds}. The \text{loud} transformation will scale the sounds by 0.25, and the \textit{stretch} (\sim) will apply to the shift (@) amounts 0.0, 0.7, 1.0, and 1.2 + \text{dly}. The sounds themselves (copies of \text{a\text{-snd}}) will not be stretched because \text{cue} never stretches sounds.

Section 7.3 describes the full set of transformations.

3.9 Overriding Default Transformations

In Nyquist, behaviors are \textit{the} important abstraction mechanism. A behavior represents a class of related functions or sounds. For example, a behavior can represent a musical note. When a note is stretched, it usually means that the tone sustains for more oscillations, but if the "note" is a drum roll, the note sustains by
more repetitions of the component drum strokes. The concept of sustain is so fundamental that we do not really think of different note durations as being different instances of an abstract behavior, but in a music programming language, we need a way to model these abstract behaviors. As the tone and drum roll examples show, there is no one right way to "stretch," so the language must allow users to define exactly what it means to stretch. By extension, the Nyquist programmer can define how all of the transformations affect different behaviors.

To make programming easier, almost all Nyquist sounds are constructed from primitive behaviors that obey the environment in obvious ways: Stretch transformations make things longer and At transformations shift things in time. But sometimes you have to override the default behaviors. Maybe the attack phase of an envelope should not stretch when the note is stretched, or maybe when you stretch a trill, you should get more notes rather than a slower trill.

To override default behaviors, you almost always follow the same programming pattern: first, capture the environment in a local variable; then, use one of the absolute transformations to "turn off" the environment’s effect and compute the sound as desired. The following example creates a very simple envelope with a fixed rise time to illustrate the technique.

```nyquist
define function two-phase-env(rise-time)
    begin
        with dur = get-duration(1)
        return pwl(rise-time, 1, dur) ~~ 1.0
    end
```

To "capture the environment in a local variable," a `with` construct is used to create the local variable `dur` and set it to the value of `get-duration(1)`, which answers the question: "If I apply use the environment to stretch something whose nominal duration is 1, what is the resulting duration?" (Since time transformations can involve continuous time deformations, this question is not as simple as it may sound, so please use the provided function rather than peeking inside the `*warp*` structure and trying to do it yourself.) Next, we "turn off" stretching using the `stretch-abs` form, which in SAL is denoted by the `~~` operator. Finally, we are ready to compute the envelope using `pwl`. Here, we use absolute durations. The first breakpoint is at `rise-time`, so the attack time is given by the `rise-time` parameter. The `pwl` decays back to zero at time `dur`, so the overall duration matches the duration expected from the environment encountered by this instance of `two-phase-env`. Note, however, that since the `pwl` is evaluated in a different environment established by `~~`, it is not stretched (or perhaps more accurately, it is stretched by 1.0). This is good because it means `rise-time` will not be stretched, but we must be careful to extend the envelope to `dur` so that it has the expected duration.

### 3.10 Sampling Rates

The global environment contains `*sound-srate*` and `*control-srate*`, which determine the sample rates of sounds and control signals. These can be overridden at any point by the transformations `sound-srate-abs` and `control-srate-abs`; for example,

```nyquist
sound-srate-abs(44100.0, osc(c4))
```
will compute a tone using a 44.1Khz sample rate even if the default rate is set to something different.

As with other components of the environment, you should *never* change *sound-srate* or *control-srate* directly. The global environment is determined by two additional variables: *default-sound-srate* and *default-control-srate*. You can add lines like the following to your `init.lsp` file to change the default global environment:

```lisp
(setf *default-sound-srate* 44100.0)
(setf *default-control-srate* 1102.5)
```

You can also do this using preferences in NyquistIDE. If you have already started Nyquist and want to change the defaults, the preferences or the following functions can be used:

```lisp
exec set-control-srate(1102.5)
exec set-sound-srate(22050.0)
```

These modify the default values and reinitialize the Nyquist environment.
Nyquist transformations were discussed in the previous chapter, but all of the examples used scalar values. For example, we saw the loud transformation used to change loudness by a fixed amount. What if we want to specify a crescendo, where the loudness changes gradually over time?

It turns out that all transformations can accept signals as well as numbers, so transformations can be continuous over time. This raises some interesting questions about how to interpret continuous transformations. Should a loudness transformation apply to the internal details of a note or only affect the initial loudness? It might seem unnatural for a decaying piano note to perform a crescendo. On the other hand, a sustained trumpet sound should probably crescendo continuously. In the case of time warping (tempo changes), it might be best for a drum roll to maintain a steady rate, a trill may or may not change rates with tempo, and a run of sixteenth notes will surely change its rate.

These issues are complex, and Nyquist cannot hope to automatically do the right thing in all cases. However, the concept of behavioral abstraction provides an elegant solution. Since transformations merely modify the environment, behaviors are not forced to implement any particular style of transformation. Nyquist is designed so that the default transformation is usually the right one, but it is always possible to override the default transformation to achieve a particular effect.

### 4.1 Simple Transformations

The "simple" transformations affect some parameter, but have no effect on time itself. The simple transformations that support continuously changing parameters are: sustain, loud, and transpose.

As a first example, Let us use transpose to create a chromatic scale. First define a sequence of tones at a steady pitch. The seqrep "function" works like seq except that it creates copies of a sound by evaluating an expression multiple times. Here, i takes on 16 values from 0 to 15, and the expression for the sound could potentially use i. Technically, seqrep is not really a function but an abbreviation for a special kind of loop construct.

```plaintext
define function tone-seq()
    return seqrep(i, 16,
                  osc-note(c4) ~ 0.25)
```

Now define a linearly increasing ramp to serve as a transposition function:
define function pitch-rise()
    return sustain-abs(1.0, 16 * ramp() ~ 4)

This ramp has a duration of 4 seconds, and over that interval it rises from 0 to 16 (corresponding to the 16 semitones we want to transpose). The ramp is inside a sustain-abs transformation, which prevents a sustain transformation from having any effect on the ramp. (One of the drawbacks of behavioral abstraction is that built-in behaviors sometimes do the wrong thing implicitly, requiring some explicit correction to turn off the unwanted transformation.) Now, pitch-rise is used to transpose tone-seq:

    define function chromatic-scale()
        return transpose(pitch-rise(), tone-seq())

Similar transformations can be constructed to change the sustain or "duty factor" of notes and their loudness. The following expression plays the chromatic-scale behavior with increasing note durations. The rhythm is unchanged, but the note length changes from staccato to legato:

    play sustain((0.2 + ramp()) ~ 4,
    chromatic-scale())

The resulting sustain function will ramp from 0.2 to 1.2. A sustain of 1.2 denotes a 20 percent overlap between notes. The sum has a stretch factor of 4, so it will extend over the 4 second duration of chromatic-scale.

If you try this, you will discover that the chromatic-scale no longer plays a chromatic scale. You will hear the first 4 notes going up in intervals of 5 semitones (perfect fourths) followed by repeated pitches. What is happening is that the sustain operation applies to pitch-rise in addition to tone-seq, so now the 4s ramp from 0 to 16 becomes a 0.8s ramp. To fix this problem, we need to shield pitch-rise from the effect of sustain using the sustain-abs transformation. Here is a corrected version of chromatic-scale:

    define function chromatic-scale()
        return transpose(sustain-abs(1, pitch-rise()), tone-seq())

What do these transformations mean? How did the system know to produce a pitch rise rather than a continuous glissando? This all relates to the idea of behavioral abstraction. It is possible to design sounds that do glissando under the transpose transform, and you can even make sounds that ignore transpose altogether. As explained in Chapter 3, the transformations modify the environment, and behaviors can reference the environment to determine what signals to generate. All built-in functions, such as osc, have a default behavior.

The default behavior for sound primitives under transpose, sustain, and loud transformations is to sample the environment at the beginning of the note. Transposition is not quantized to semitones or any other scale, but in our example, we arranged for the transposition to work out to integer numbers of semitones, so we obtained a chromatic scale anyway.

Transposition only applies to the oscillator and sampling primitives osc, partial, sampler, sine, fmosc, and amosc. Sustain applies to osc, env, ramp, and pwl. (Note that partial, amosc, and fmosc get their durations from the modulation signal, so they may indirectly depend upon the sustain.) Loud applies to osc, sampler, cue, sound, fmosc, and amosc. (But not pwl or env.)

v 3.12
4.2 Time Warps

The most interesting transformations have to do with transforming time itself. The warp transformation provides a mapping function from logical (score) time to real time. The slope of this function tells us how many units of real time are covered by one unit of score time. This is proportional to 1/tempo. A higher slope corresponds to a slower tempo.

To demonstrate warp, we will define a time warp function using pwl:

```lisp
define function warper()
    return pwl(0.25, .4, .75, .6, 1.0, 1.0, 2.0, 2.0, 2.0)
```

This function has an initial slope of \( \frac{.4}{.25} = 1.6 \). It may be easier to think in reciprocal terms: the initial tempo is \( \frac{.25}{.4} = .625 \). Between 0.25 and 0.75, the tempo is \( \frac{.5}{.2} = 2.5 \), and from 0.75 to 1.0, the tempo is again \( .625 \). It is important for warp functions to completely span the interval of interest (in our case it will be 0 to 1), and it is safest to extend a bit beyond the interval, so we extend the function on to 2.0 with a tempo of 1.0. Next, we stretch and scale the warper function to cover 4 seconds of score time and 4 seconds of real time:

```lisp
define function warp4()
    return 4 * warper() ~ 4
```

Figure 4.1 shows a plot of this warp function. Now, we can warp the tempo of the tone-seq defined above using warp4:

```lisp
play warp(warp4(), tone-seq())
```

Figure 4.2 shows the result graphically. Notice that the durations of the tones are warped as well as their onsets. Envelopes are not shown in detail in the figure. Because of the way env is defined, the tones will have constant attack and decay times, and the sustain will be adjusted to fit the available time.

4.3 Abstract Time Warps

We have seen a number of examples where the default behavior did the "right thing," making the code straightforward. This is not always the case. Suppose we want to warp the note onsets but not the durations. We will first look at an incorrect solution and discuss the error. Then we will look at a slightly more complex (but correct) solution.

The default behavior for most Nyquist built-in functions is to sample the time warp function at the nominal starting and ending score times of the primitive. For many built-in functions, including osc, the starting logical time is 0 and the ending logical time is 1, so the time warp function is evaluated at these points to yield real starting and stopping times, say 15.23 and 16.79. The difference (e.g. 1.56) becomes the signal duration, and there is no internal time warping. The pwl function behaves a little differently. Here, each breakpoint is warped individually, but the resulting function is linear between the breakpoints.
Figure 4.1: The result of \((\text{warp4})\), intended to map 4 seconds of score time into 4 seconds of real time. The function extends beyond 4 seconds (the dashed lines) to make sure the function is well-defined at location \((4, 4)\). Nyquist sounds are ordinarily open on the right.

Figure 4.2: When \((\text{warp4})\) is applied to \((\text{tone-seq-2})\), the note onsets and durations are warped.
Figure 4.3: When \( \text{warp4} \) is applied to \( \text{tone-seq-3} \), the note onsets are warped, but not the duration, which remains a constant 0.25 seconds. In the fast middle section, this causes notes to overlap. Nyquist will sum (mix) them.

A consequence of the default behavior is that notes stretch when the tempo slows down. Returning to our example, recall that we want to warp only the note onset times and not the duration. One would think that the following would work:

```
define function tone-seq-2 ()
  return seqrep(i, 16,  
              osc-note(c4) ~~ 0.25)
```

```
play warp(warp4(), tone-seq-2())
```

Here, we have redefined \textbf{tone-seq}, renaming it to \textbf{tone-seq-2} and changing the stretch (\( \sim \)) to absolute stretch (\( \sim\sim \)). The absolute stretch should override the warp function and produce a fixed duration.

If you play the example, you will hear steady sixteenths and no tempo changes. What is wrong? In a sense, the "fix" works too well. Recall that sequences (including \textbf{seqrep}) determine the starting time of the next note from the logical stop time of the previous sound in the sequence. When we forced the stretch to 0.25, we also forced the logical stop time to 0.25 real seconds from the beginning, so every note starts 0.25 seconds after the previous one, resulting in a constant tempo.

Now let us design a proper solution. The trick is to use absolute stretch (\( \sim\sim \)) as before to control the duration, but to restore the logical stop time to a value that results in the proper inter-onset time interval:

```
define function tone-seq-3 ()
  return seqrep(i, 16,  
              set-logical-stop(osc-note(c4) \( \sim\sim \) 0.25, 0.25))
```

```
play warp(warp4(), tone-seq-3())
```

Notice the addition of \textbf{set-logical-stop} enclosing the absolute stretch (\( \sim\sim \)) expression to set the logical stop time. A possible point of confusion here is that the logical stop time is set to 0.25, the same number given to \( \sim\sim \)! How does setting the logical stop time to 0.25 result in a tempo change? When used within a \textbf{warp} transformation, the second argument to \textbf{set-logical-stop} refers to \textit{score} time rather than \textit{real} time. Therefore, the score duration of 0.25 is warped into real time, producing tempo changes according to the environment. Figure 4.3 illustrates the result graphically.
4.4 Nested Transformations

Transformations can be nested. In particular, a simple transformation such as transpose can be nested within a time warp transformation. Suppose we want to warp our chromatic scale example with the warp4 time warp function. As in the previous section, we will show an erroneous simple solution followed by a correct one.

The simplest approach to a nested transformation is to simply combine them and hope for the best:

```plaintext
play warp(warp4(),
    transpose(pitch-rise(), tone-seq()))
```

This example will not work the way you might expect. Here is why: the warp transformation applies to the (pitch-rise) expression, which is implemented using the ramp function. The default behavior of ramp is to interpolate linearly (in real time) between two points. Thus, the "warped" ramp function will not truly reflect the internal details of the intended time warp. When the notes are moving faster, they will be closer together in pitch, and the result is not chromatic. What we need is a way to properly compose the warp and ramp functions. If we continuously warp the ramp function in the same way as the note sequence, a chromatic scale should be obtained. This will lead to a correct solution.

Here is the modified code to properly warp a transposed sequence. Note that the original sequence is used without modification. The only complication is producing a properly warped transposition function:

```plaintext
play warp(warp4(),
    transpose(
        control-warp(get-warp(),
            warp-abs(nil, pitch-rise())),
        tone-seq()))
```

To properly warp the pitch-rise transposition function, we use control-warp, which applies a warp function to a function of score time, yielding a function of real time. We need to pass the desired function to control-warp, so we fetch it from the environment with get-warp(). Finally, since the warping is done here, we want to shield the pitch-rise expression from further warping, so we enclose it in warp-abs(nil, ...).

An aside: This last example illustrates a difficulty in the design of Nyquist. To support behavioral abstraction universally, we must rely upon behaviors to "do the right thing." In this case, we would like the ramp function to warp continuously according to the environment. But this is inefficient and unnecessary in many other cases where ramp and especially pwl are used. (pwl warps its breakpoints, but still interpolates linearly between them.) Also, if the default behavior of primitives is to warp in a continuous manner, this makes it difficult to build custom abstract behaviors. The final vote is not in.
5 More Examples

This chapter explores Nyquist through additional examples. The reader may wish to browse through these and move on to Chapter 7, which is a reference section describing Nyquist functions.

5.1 Stretching Sampled Sounds

This example illustrates how to stretch a sound, resampling it in the process. Because sounds in Nyquist are values that contain the sample rate, start time, etc., use sound to convert a sound into a behavior that can be stretched, e.g. sound(a-snd). This behavior stretches a sound according to the stretch factor in the environment, set using stretch. For accuracy and efficiency, Nyquist does not resample a stretched sound until absolutely necessary. The force-srate function is used to resample the result so that we end up with a "normal" sample rate that is playable on ordinary sound cards.

```nyquist
; if a-snd is not loaded, load sound sample:
;
if not(boundp(quote(a-snd))) then
  set a-snd = s-read("demo-snd.aiff")

; the SOUND operator shifts, stretches, clips and scales
; a sound according to the current environment
;
define function ex23()
  play force-srate(*default-sound-srate*, sound(a-snd) ~ 3.0)

define function down()
  return force-srate(*default-sound-srate*,
                      seq(sound(a-snd) ~ 0.2,
                          sound(a-snd) ~ 0.3,
                          sound(a-snd) ~ 0.4,
                          sound(a-snd) ~ 0.6))

play down()
```
Nyquist Reference Manual

5. More Examples

; that was so much fun, let's go back up:
;
define function up()
    return force-srate(*default-sound-srate*,
        seq(sound(a-snd) ~ 0.5,
            sound(a-snd) ~ 0.4,
            sound(a-snd) ~ 0.3,
            sound(a-snd) ~ 0.2))

; and write a sequence
;
play seq(down(), up(), down())

Notice the use of the sound behavior as opposed to cue. The cue behavior shifts and scales its sound according to *warp* and *loud*, but it does not change the duration or resample the sound. In contrast, sound not only shifts and scales its sound, but it also stretches it by resampling or changing the effective sample rate according to *warp*. If *warp* is a continuous warping function, then the sound will be stretched by time-varying amounts. (The *transpose* element of the environment is ignored by both cue and sound.)

Note: sound may use linear interpolation rather than a high-quality resampling algorithm. In some cases, this may introduce errors audible as noise. Use resample (see Section 7.2.2) for high-quality interpolation.

In the functions up and down, the *warp* is set by stretch (~), which simply scales time by a constant scale factor. In this case, sound can "stretch" the signal simply by changing the sample rate without any further computation. When seq tries to add the signals together, it discovers the sample rates do not match and uses linear interpolation to adjust all sample rates to match that of the first sound in the sequence. The result of seq is then converted using force-srate to convert the sample rate, again using linear interpolation. It would be slightly better, from a computational standpoint, to apply force-srate individually to each stretched sound rather than applying force-srate after seq.

Notice that the overall duration of sound(a-snd) ~ 0.5 will be half the duration of a-snd.

5.2 Saving Sound Files

So far, we have used the play command to play a sound. The play command works by writing a sound to a file while simultaneously playing it. This can be done one step at a time, and it is often convenient to save a sound to a particular file for later use:

; write the sample to a file,
; the file name can be any Unix filename. Prepending a "./" tells
; s-save to not prepend *default-sf-dir*
;
exec s-save(a-snd, 1000000000, "./a-snd-file.snd")
; play a file
; play command normally expects an expression for a sound
; but if you pass it a string, it will open and play a
; sound file
play "./a-snd-file.snd"

; delete the file (do this with care!)
; only works under Unix (not Windows)
exec system("rm ./a-snd-file.snd")

; now let's do it using a variable as the file name
;
set my-sound-file = "/a-snd-file.snd"
exec s-save(a-snd, 1000000000, my-sound-file)

; play-file is a function to open and play a sound file
exec play-file(my-sound-file)
exec system(strcat("rm ", my-sound-file))

This example shows how \texttt{s-save} can be used to save a sound to a file.

The last line of this example shows how the \texttt{system} function can be used to invoke Unix shell commands, such as a command to play a file or remove it. Finally, notice that \texttt{strcat} can be used to concatenate a command name to a file name to create a complete command that is then passed to \texttt{system}. (This is convenient if the sound file name is stored in a parameter or variable.)

5.3 Memory Space and Normalization

Sound samples take up lots of memory, and often, there is not enough primary (RAM) memory to hold a complete composition. For this reason, Nyquist can compute sounds incrementally, saving the final result on disk. However, Nyquist can also save sounds in memory so that they can be reused efficiently. In general, if a sound is saved in a global variable, memory will be allocated as needed to save and reuse it.

The standard way to compute a sound and write it to disk is to pass an expression to the \texttt{play} command:

\begin{verbatim}
play my-composition()
\end{verbatim}

Often it is nice to normalize sounds so that they use the full available dynamic range of 16 bits. Nyquist has an automated facility to help with normalization. By default, Nyquist computes up to 1 million samples (using about 4MB of memory) looking for the peak. The entire sound is normalized so that this peak will not cause clipping. If the sound has less than 1 million samples, or if the first million samples are a good indication of the overall peak, then the signal will not clip.

With this automated normalization technique, you can choose the desired peak value by setting \texttt{*autonorm-target*}, which is initialized to 0.9. The number of samples examined is \texttt{*autonorm-max-samples*}, initially 1 million. You can turn this feature off by executing:
exec autonorm-off()

and turn it back on by typing:

exec autonorm-on()

This normalization technique is in effect when *autonorm-type* is quote(lookahead), which is the default.

An alternative normalization method uses the peak value from the previous call to play. After playing a file, Nyquist can adjust an internal scale factor so that if you play the same file again, the peak amplitude will be *autonorm-target*, which is initialized to 0.9. This can be useful if you want to carefully normalize a big sound that does not have its peak near the beginning. To select this style of normalization, set *autonorm-type* to the (quoted) atom quote(previous).

You can also create your own normalization method in Nyquist. The peak function computes the maximum value of a sound. The peak value is also returned from the play macro. You can normalize in memory if you have enough memory; otherwise you can compute the sound twice. The two techniques are illustrated here:

; normalize in memory. First, assign the sound to a variable so
; it will be retained:
set mysound = sim(osc(c4), osc(c5))
; now compute the maximum value (ny:all is 1 giga-samples, you may want a
; smaller constant if you have less than 4GB of memory:
set mymax = snd-max(mysound, NY:ALL)
display "Computed max", mymax
; now write out and play the sound from memory with a scale factor:
play mysound * (0.9 / mymax)

; if you don't have space in memory, here's how to do it:
define function myscore()
  return sim(osc(c4), osc(c5))
; compute the maximum:
set mymax = snd-max(list(quote(myscore)), NY:ALL)
display "Computed max", mymax
; now we know the max, but we don't have a the sound (it was garbage
; collected and never existed all at once in memory). Compute the sound
; again, this time with a scale factor:
play myscore() * (0.9 / mymax)

You can also write a sound as a floating point file. This file can then be converted to 16-bit integer with the proper scaling applied. If a long computation was involved, it should be much faster to scale the saved sound file than to recompute the sound from scratch. Although not implemented yet in Nyquist, some header formats can store maximum amplitudes, and some soundfile player programs can rescale floating point files on the fly, allowing normalized soundfile playback without an extra normalization pass (but at a cost of twice the disk space of 16-bit samples). You can use Nyquist to rescale a floating point file and convert it to 16-bit samples for playback.
5.4 Frequency Modulation

The next example uses the Nyquist frequency modulation behavior `fmosc` to generate various sounds. The parameters to `fmosc` are:

```
fmosc(pitch, modulator, table, phase)
```

Note that pitch is the number of half-steps, e.g. `c4` has the value of 60 which is middle-C, and phase is in degrees. Only the first two parameters are required:

```
; make a short sine tone with no frequency modulation
; play fmosc(c4, pwl(0.1))
```

```
; make a longer sine tone -- note that the duration of
; the modulator determines the duration of the tone
; play fmosc(c4, pwl(0.5))
```

In the example above, `pwl` (for Piece-Wise Linear) is used to generate sounds that are zero for the durations of 0.1 and 0.5 seconds, respectively. In effect, we are using an FM oscillator with no modulation input, and the result is a sine tone. The duration of the modulation determines the duration of the generated tone (when the modulation signal ends, the oscillator stops).

The next example uses a more interesting modulation function, a ramp from zero to `C4`, expressed in hz. More explanation of `pwl` is in order. This operation constructs a piece-wise linear function sampled at the `*control-srate*`. The first breakpoint is always at `(0, 0)`, so the first two parameters give the time and value of the second breakpoint, the second two parameters give the time and value of the third breakpoint, and so on. The last breakpoint has a value of 0, so only the time of the last breakpoint is given. In this case, we want the ramp to end at `C4`, so we cheat a bit by having the ramp return to zero "almost" instantaneously between times 0.5 and 0.501.

The `pwl` behavior always expects an odd number of parameters. The resulting function is shifted and stretched linearly according to `*warp*` in the environment. Now, here is the example:

```
; make a frequency sweep of one octave; the piece-wise linear function
; sweeps from 0 to (step-to-hz c4) because, when added to the c4
; fundamental, this will double the frequency and cause an octave sweep.
; play fmosc(c4, pwl(0.5, step-to-hz(c4), 0.501))
```

The same idea can be applied to a non-sinusoidal carrier. Here, we assume that `*fm-voice*` is predefined (the next section shows how to define it):

```
; do the same thing with a non-sine table
; play fmosc(cs2, pwl(0.5, step-to-hz(cs2), 0.501),
  *fm-voice*, 0.0)
```
The next example shows how a function can be used to make a special frequency modulation contour. In this case the contour generates a sweep from a starting pitch to a destination pitch:

```nyquist
; make a function to give a frequency sweep, starting
; after <delay> seconds, then sweeping from <pitch-1>
; to <pitch-2> in <sweep-time> seconds and then
; holding at <pitch-2> for <hold-time> seconds.
;
define function sweep(delay, pitch-1, sweep-time, pitch-2, hold-time)
    begin
        with interval = step-to-hz(pitch-2) - step-to-hz(pitch-1)
        return pwl(delay, 0.0,
            ; sweep from pitch 1 to pitch 2
            delay + sweep-time, interval,
            ; hold until about 1 sample from the end
            delay + sweep-time + hold-time - 0.0005, interval,
            ; quickly ramp to zero (pwl always does this,
            ; so make it short)
            delay + sweep-time + hold-time)
    end
```

; now try it out
;
play fmosc(cs2, sweep(0.1, cs2, 0.6, gs2, 0.5), *fm-voice*, 0.0)

FM can be used for vibrato as well as frequency sweeps. Here, we use the lfo function to generate vibrato. The lfo operation is similar to osc, except it generates sounds at the *control-srate*, and the parameter is hz rather than a pitch:

```nyquist
play fmosc(cs2, 10.0 * lfo(6.0), *fm-voice*, 0.0)
```

What kind of manual would this be without the obligatory FM sound? Here, a sinusoidal modulator (frequency C₄) is multiplied by a slowly increasing ramp from zero to 1000.0.

```nyquist
set modulator = pwl(1.0, 1000.0, 1.0005) * osc(c4)
    ; make the sound
play fmosc(c4, modulator)
```

For more simple examples of FM in Nyquist, see demos/warble_tutorial.htm. Another interesting FM sound reminiscent of "scratching" can be found with a detailed explanation in demos/scratch_tutorial.htm.
5.5 Building a Wavetable

In Section 1.7.1, we saw how to synthesize a wavetable. A wavetable for osc also can be extracted from any sound. This is especially interesting if the sound is digitized from some external sound source and loaded using the s-read function. Recall that a table is a list consisting of a sound, the pitch of that sound, and T (meaning the sound is periodic).

In the following, a sound is first read from the file demo-snd.nh. Then, the extract function is used to extract the portion of the sound between 0.110204 and 0.13932 seconds. (These numbers might be obtained by first plotting the sound and estimating the beginning and end of a period, or by using some software to look for good zero crossings.) The result of extract becomes the first element of a list. The next element is the pitch (24.848422), and the last element is T. The list is assigned to *fm-voice*.

```lisp
if not(boundp(quote(a-snd))) then
  set a-snd = s-read("demo-snd.aiff")

set *fm-voice* = list(extract(0.110204, 0.13932, cue(a-snd)),
  24.848422,
  #T)
```

The file demos/examples.sal contains an extensive example of how to locate zero-crossings, extract a period, build a waveform, and generate a tone from it. (See ex37 through ex40 in the file.)

5.6 Filter Examples

Nyquist provides a variety of filters. All of these filters take either real numbers or signals as parameters. If you pass a signal as a filter parameter, the filter coefficients are recomputed at the sample rate of the control signal. Since filter coefficients are generally expensive to compute, you may want to select filter control rates carefully. Use control-srate-abs (Section 7.3) to specify the default control sample rate, or use force-srate (Section 7.2.2) to resample a signal before passing it to a filter.

Before presenting examples, let’s generate some unfiltered white noise:

```lisp
play noise()
```

Now low-pass filter the noise with a 1000Hz cutoff:

```lisp
play lp(noise(), 1000.0)
```

The high-pass filter is the inverse of the low-pass:

```lisp
play hp(noise(), 1000.0)
```

Here is a low-pass filter sweep from 100Hz to 2000Hz:

```lisp
play lp(noise(), pwl(0.0, 100.0, 1.0, 2000.0, 1.0))
```

And a high-pass sweep from 50Hz to 4000Hz:
play hp(noise(), pwl(0.0, 50.0, 1.0, 4000.0, 1.0))

The band-pass filter takes a center frequency and a bandwidth parameter. This example has a 500Hz center frequency with a 20Hz bandwidth. The scale factor is necessary because, due to the resonant peak of the filter, the signal amplitude exceeds 1.0:

play reson(10.0 * noise(), 500.0, 20.0, 1)

In the next example, the center frequency is swept from 100 to 1000Hz, using a constant 20Hz bandwidth:

play reson(0.04 * noise(), pwl(0.0, 200.0, 1.0, 1000.0, 1.0), 20.0)

For another example with explanations, see demos/wind_tutorial.htm.

5.7 DSP in Lisp

In almost any signal processing system, the vast majority of computation takes place in the inner loops of DSP algorithms, and Nyquist is designed so that these time-consuming inner loops are in highly-optimized machine code rather than relatively slow interpreted lisp code. As a result, Nyquist typically spends 95% of its time in these inner loops; the overhead of using a Lisp interpreter is negligible.

The drawback is that Nyquist must provide the DSP operations you need, or you are out of luck. When Nyquist is found lacking, you can either write a new primitive signal operation, or you can perform DSP in Lisp code. Neither option is recommended for inexperienced programmers. Instructions for extending Nyquist are given in Appendix A. This section describes the process of writing a new signal processing function in Lisp.

Before implementing a new DSP function, you should decide which approach is best. First, figure out how much of the new function can be implemented using existing Nyquist functions. For example, you might think that a tapped-delay line would require a new function, but in fact, it can be implemented by composing sound transformations to accomplish delays, scale factors for attenuation, and additions to combine the intermediate results. This can all be packaged into a new Lisp function, making it easy to use. If the function relies on built-in DSP primitives, it will execute very efficiently.

Assuming that built-in functions cannot be used, try to define a new operation that will be both simple and general. Usually, it makes sense to implement only the kernel of what you need, combining it with existing functions to build a complete instrument or operation. For example, if you want to implement a physical model that requires a varying breath pressure with noise and vibrato, plan to use Nyquist functions to add a basic pressure envelope to noise and vibrato signals to come up with a composite pressure signal. Pass that signal into the physical model rather than synthesizing the envelope, noise, and vibrato within the model. This not only simplifies the model, but gives you the flexibility to use all of Nyquist’s operations to synthesize a suitable breath pressure signal.

Having designed the new "kernel" DSP operation that must be implemented, decide whether to use C or Lisp. (At present, SAL is not a good option because it has no support for object-oriented programming.) To
use C, you must have a C compiler, the full source code for Nyquist, and you must learn about extending Nyquist by reading Appendix A. This is the more complex approach, but the result will be very efficient. A C implementation will deal properly with sounds that are not time-aligned or matched in sample rates. To use Lisp, you must learn something about the XLISP object system, and the result will be about 50 times slower than C. Also, it is more difficult to deal with time alignment and differences in sample rates. The remainder of this section gives an example of a Lisp version of \texttt{snd-prod} to illustrate how to write DSP functions for Nyquist in Lisp.

The \texttt{snd-prod} function is the low-level multiply routine. It has two sound parameters and returns a sound which is the product of the two. To keep things simple, we will assume that two sounds to be multiplied have a matched sample rate and matching start times. The DSP algorithm for each output sample is simply to fetch a sample from each sound, multiply them, and return the product.

To implement \texttt{snd-prod} in Lisp, three components are required:

1. An object is used to store the two parameter sounds. This object will be called upon to yield samples of the result sound;
2. Within the object, the \texttt{snd-fetch} routine is used to fetch samples from the two input sounds as needed;
3. The result must be of type \texttt{SOUND}, so \texttt{snd-fromobject} is used to create the result sound.

The combined solution will work as follows: The result is a value of type \texttt{sound} that retains a reference to the object. When Nyquist needs samples from the sound, it invokes the sound's "fetch" function, which in turn sends an XLISP message to the object. The object will use \texttt{snd-fetch} to get a sample from each stored sound, multiply the samples, and return a result.

Thus the goal is to design an XLISP object that, in response to a \texttt{:next} message will return a proper sequence of samples. When the sound reaches the termination time, simply return \texttt{NIL}.

The XLISP manual (see Appendix C) describes the object system, but in a very terse style, so this example will include some explanation of how the object system is used. First, we need to define a class for the objects that will compute sound products. Every class is a subclass of class \texttt{class}, and you create a subclass by sending \texttt{:new} to a class.

\begin{verbatim}
(setf product-class (send class :new '(s1 s2)))
\end{verbatim}

The parameter \texttt{'(s1 s2)} says that the new class will have two instance variables, \texttt{s1} and \texttt{s2}. In other words, every object which is an instance of class \texttt{product-class} will have its own copy of these two variables.

Next, we will define the \texttt{:next} method for \texttt{product-class}:

\begin{verbatim}
(send product-class :answer :next '()
 '((let ((f1 (snd-fetch s1))
        (f2 (snd-fetch s2)))
   (cond ((and f1 f2)
          (* f1 f2))
         (t nil))))))
\end{verbatim}
The :answer message is used to insert a new method into our new product-class. The method is described in three parts: the name (:next), a parameter list (empty in this case), and a list of expressions to be evaluated. In this case, we fetch samples from s1 and s2. If both are numbers, we return their product. If either is NIL, we terminate the sound by returning nil.

The :next method assumes that s1 and s2 hold the sounds to be multiplied. These must be installed when the object is created. Objects are created by sending :new to a class. A new object is created, and any parameters passed to :new are then sent in a :isnew message to the new object. Here is the :isnew definition for product-class:

```lisp
(send product-class :answer :isnew 'p1 p2
  '((setf s1 (snd-copy p1))
   (setf s2 (snd-copy p2))))
```

Take careful note of the use of snd-copy in this initialization. The sounds s1 and s2 are modified when accessed by snd-fetch in the :next method defined above, but this destroys the illusion that sounds are immutable values. The solution is to copy the sounds before accessing them; the original sounds are therefore unchanged. (This copy also takes place implicitly in most Nyquist sound functions.)

To make this code safer for general use, we should add checks that s1 and s2 are sounds with identical starting times and sample rates; otherwise, an incorrect result might be computed.

Now we are ready to write snd-product, an approximate replacement for snd-prod:

```lisp
(defun snd-product (s1 s2)
  (let (obj)
    (setf obj (send product-class :new s1 s2))
    (snd-fromobject (snd-t0 s1) (snd-srate s1) obj)))
```

This code first creates obj, an instance of product-class, to hold s1 and s2. Then, it uses obj to create a sound using snd-fromobject. This sound is returned from snd-product. Note that in snd-fromobject, you must also specify the starting time and sample rate as the first two parameters. These are copied from s1, again assuming that s1 and s2 have matching starting times and sample rates.

Note that in more elaborate DSP algorithms we could expect the object to have a number of instance variables to hold things such as previous samples, waveform tables, and other parameters.
Nyquist supports two languages: XLISP and SAL. In some sense, XLISP and SAL are the same language, but with differing syntax. This chapter describes SAL: how it works, SAL syntax and semantics, and the relationship between SAL and XLISP, and differences between Nyquist SAL and Common Music SAL.

Nyquist SAL is based on Rick Taube’s SAL language, which is part of Common Music. SAL offers the power of Lisp but features a simple, Algol-like syntax. SAL is implemented in Lisp: Lisp code translates SAL into a Lisp program and uses the underlying Lisp engine to evaluate the program. Aside from the translation time, which is quite fast, SAL programs execute at about the same speed as the corresponding Lisp program. (Nyquist SAL programs run just slightly slower than XLISP because of some runtime debugging support automatically added to user programs by the SAL compiler.)

From the user’s perspective, these implementation details are hidden. You can enter SAL mode from XLISP by typing (SAL) to the XLISP prompt. The SAL input prompt (SAL> ) will be displayed. From that point on, you simply type SAL commands, and they will be executed. By setting a preference in the NyquistIDE program, SAL mode will be entered automatically.

It is possible to encounter errors that will take you from the SAL interpreter to an XLISP prompt. In general, the way to get back to SAL is by typing (top) to get back to the top level XLISP interpreter and reset the Nyquist environment. Then type (sal) to restart the SAL interpreter.

### 6.1 SAL Syntax and Semantics

The most unusual feature of SAL syntax is that identifiers are Lisp-like, including names such as "play-file" and even "*warp*." In SAL, most operators must be separated from identifiers by white space. For example, play-file is one identifier, but play - file is an expression for "play minus file," where play and file are two separate identifiers. Fortunately, no spaces are needed around commas and parentheses.

The set SAL identifiers is difficult to describe due to a number of interacting rules designed to prevent surprising or confusing code. The exact details can be found in the symbol-token? function in sal-parse.lsp. To determine if an identifier is allowed, you can also follow the following example, which tests whether $a@$ is a valid identifier:

```
SAL> print type-of(quote($a@))
```
If this statement prints SYMBOL, then the quoted expression is a valid identifier. If SAL reports a "parse error" or some other type, then the quoted expression is not a valid identifier.

In SAL, whitespace (any sequence of space, newline, or tab characters) is sometimes necessary to separate lexical tokens, but otherwise, spaces and indentation are ignored. To make SAL readable, it is strongly advised that you indent SAL programs as in the examples here. The NyquistIDE program is purposely insistent about SAL indentation, so if you use it to edit SAL programs, your indentation should be both beautiful and consistent.

As in Lisp (but very unlike C or Java), comments are indicated by semicolons. Any text from an unquoted colon to the end of the line is ignored.

```sal
; this is a comment
; comments are ignored by the compiler
print "Hello World" ; this is a SAL statement
```

As in Lisp, identifiers are translated to upper-case, making SAL case-insensitive. For example, the function name autonorm can be typed in lower case or as AUTONORM, AutoNorm, or even AuToNoRn. All forms denote the same function. The recommended approach is to write programs in all lower case.

SAL is organized around statements, most of which contain expressions. We will begin with expressions and then look at statements.

### 6.1.1 Expressions

#### 6.1.1.1 Simple Expressions

As in XLISP, simple expressions include:

- integers (FIXNUM’s), such as 1215,
- floats (FLONUM’s) such as 12.15,
- strings (STRING’s) such as "Magna Carta", and
- symbols (SYMBOL’s) such as magna-carta. A symbol with a leading colon (:) evaluates to itself as in Lisp. Otherwise, a symbol denotes either a local variable, a formal parameter, or a global variable. As in Lisp, variables do not have data types or type declarations. The type of a variable is determined at runtime by its value.

Additional simple expressions in SAL are:

- lists such as `{c 60 e 64}`. Note that there are no commas to separate list elements, and symbols in lists are not evaluated as variables but stand for themselves. Lists may contain numbers, booleans (which represent XLisp’s `T` or `NIL`, SAL identifiers (representing XLisp symbols), strings, SAL operators (representing XLisp symbols), and nested lists.
• Booleans: SAL interprets #t as true and #f as false. (As far as the SAL compiler is concerned, t and nil are just variables. Since these are the Lisp versions of true and false, they are interchangeable with #t and #f, respectively.)

A curious property of Lisp and Sal is that false and the empty list are the same value. Since SAL is based on Lisp, #f and {} (the empty list) are equal.

### 6.1.1.2 Operators

Expressions can be formed with unary and binary operators using infix notation. The operators are:

+ **addition**, including sounds
- **subtraction**, including sounds
* **multiplication**, including sounds
/ **division** (due to divide-by-zero problems, does not operate on sounds)
% **modulus** (remainder after division)
^ **exponentiation**
= **equal** (using Lisp equal for non-lists, and comparing lists element-by-element recursively)\(^1\)
!= **not equal**
> **greater than**
< **less than**
>= **greater than or equal**

---

\(^1\)In the Common Music implementation of SAL, = compiles to the Lisp = function, so this is an incompatible difference.
<=  

less than or equal

~=  

approximately equal. Numbers are approximately equal if they are within \(*\~tolerance*\) of each other. \(*\~tolerance*\) is initially 0.000001. Non-numbers are compared with the XLISP \(equal\) function, and lists are compared element-by-element (recursively) using \(~=\).\(^2\)

&  

logical and

|  

logical or

!  

logical not (unary)

<>  

time shift

@@  

time shift to absolute time

~  

time stretch

~~  

time stretch to absolute stretch factor

Again, remember that operators must be delimited from their operands using spaces or parentheses. Operator precedence is based on the following levels of precedence:

\[
\begin{align*}
\& & \@ & & \@@ & & \sim & & \sim\sim \\
/ & \times & & \div & \div & \sim & \sim & <= & >= & \approx & = \\
! & & & \& & & & \| & & & & \\
\end{align*}
\]

\(^2\)In the Common Music implementation of SAL, \(~=\) compiles to the Lisp \(equal\) function, so this is an incompatible difference.
6.1.1.3 Function Calls

A function call is a function name followed by zero or more comma-delimited argument expressions enclosed within parentheses:

- list()
- piano-note(2.0, c4 + interval, 100)

Some functions use named parameters, in which case the name of the argument with a colon precedes the argument expression.

- s-save(my-snd(), ny:all, "tmp.wav", play: #t, bits: 16)

6.1.1.4 Array Notation

An array reference is a variable identifier followed by an index expression in square brackets, e.g.:

- x[23] + y[i]

6.1.1.5 Conditional Values

The special operator #? evaluates the first argument expression. If the result is true, the second expression is evaluated and its value is returned. If false, the third expression is evaluated and returned (or false is returned if there is no third expression):

- #?(random(2) = 0, unison, major-third)
- #?(pitch >= c4, pitch - c4); returns false if pitch < c4

6.1.2 SAL Statements

SAL compiles and evaluates statements one at a time. You can type statements at the SAL prompt or load a file containing SAL statements. SAL statements are described below. The syntax is indicated at the beginning of each statement type description: this font indicates literal terms such as keywords, the italic font indicates a place-holder for some other statement or expression. Bracket [like this] indicate optional (zero or one) syntax elements, while braces with a plus {like this}+ indicate one or more occurrences of a syntax element. Braces with a star {like this}* indicate zero or more occurrences of a syntax element: {non-terminal}* is equivalent to [ {non-terminal}+ ].

6.1.2.1 begin and end

begin [with-stmt] {statement}+ end

A begin-end statement consists of a sequence of statements surrounded by the begin and end keywords. This form is often used for function definitions and after then or else where the syntax demands a single
statement but you want to perform more than one action. Variables may be declared using an optional with
statement immediately after begin. For example:

begin
  with db = 12.0,
  linear = db-to-linear(db)
  print db, "dB represents a factor of", linear
  set scale-factor = linear
end

6.1.2.2 chdir

chdir expression

The chdir statement changes the working directory. This statement is provided for compatibility with
Common Music SAL, but it really should be avoided if you use NyquistIDE. The expression following the
chdir keyword should evaluate to a string that is a directory path name. Note that literal strings themselves
are valid expressions.

chdir "/Users/rbd/tmp"

6.1.2.3 define variable

[define] variable name [= expression] {, name [= expression]}*

Global variables can be declared and initialized. A list of variable names, each with an optional initialization
follows the define variable keywords. (Since variable is a keyword, define is redundant and optional
in Nyquist SAL, but required in Common Music SAL.) If the initialization part is omitted, the variable is
initialized to false. Global variables do not really need to be declared: just using the name implicitly creates
the corresponding variable. However, it is an error to use a global variable that has not been initialized;
define variable is a good way to introduce a variable (or constant) with an initial value into your program.

define variable transposition = 2,
  print-debugging-info, ; initially false
  output-file-name = "salmon.wav"

6.1.2.4 define function

[define] function name ([parameter] {, parameter}*) statement

Before a function be called from an expression (as described above), it must be defined. A function definition
gives the function name, a list of parameters, and a statement. When a function is called, the actual parameter
expressions are evaluated from left to right and the formal parameters of the function definition are set to
these values. Then, statement is evaluated.
The formal parameters may be positional parameters that are matched with actual parameters by position from left to right. Syntactically, these are symbols and these symbols are essentially local variables that exist only until statement completes or a return statement causes the function evaluation to end. As in Lisp, parameters are passed by value, so assigning a new value to a formal parameter has no effect on the actual value. However, lists and arrays are not copied, so internal changes to a list or array produce observable side effects.

Alternatively, formal parameters may be keyword parameters. Here the parameter is actually a pair: a keyword parameter, which is a symbol followed by a colon, and a default value, given by any expression. Within the body of the function, the keyword parameter is named by a symbol whose name matches the keyword parameter except there is no final colon.

```
define function foo(x: 1, y: bar(2, 3))
    display "foo", x, y

exec foo(x: 6, y: 7)
```

In this example, x is bound to the value 6 and y is bound to the value 7, so the example prints "foo : X = 6, Y = 7". Note that while the keyword parameters are x: and y:, the corresponding variable names in the function body are x and y, respectively.

The parameters are meaningful only within the lexical (static) scope of statement. They are not accessible from within other functions even if they are called by this function.

Use a begin-end statement if the body of the function should contain more than one statement or you need to define local variables. Use a return statement to return a value from the function. If statement completes without a return, the value false is returned.

6.1.2.5 exec

```
exec expression
```

Unlike most other programming languages, you cannot simply type an expression as a statement. If you want to evaluate an expression, e.g. call a function, you must use an exec statement. The statement simply evaluates the expression. For example,

```
exec set-sound-srate(22050.0) ; change default sample rate
```

6.1.2.6 if

```
if test-expr then true-stmt [else false-stmt]
```

An if statement evaluates the expression test-expr. If it is true, it evaluates the statement true-stmt. If false, the statement false-stmt is evaluated. Use a begin-end statement to evaluate more than one statement in then then or else parts.
if \( x < 0 \) then \( x = -x \); \( x \) gets its absolute value

if \( x > \text{upper-bound} \) then
begin
print "x too big, setting to", upper-bound
\( x = \text{upper-bound} \)
end
else
if \( x < \text{lower-bound} \) then
begin
print "x too small, setting to", lower-bound
\( x = \text{lower-bound} \)
end

Notice in this example that the else part is another if statement. An if may also be the then part of another if, so there could be two possible if's with which to associate an else. An else clause always associates with the closest previous if that does not already have an else clause.

6.1.2.7 when

when test statement

The when statement is similar to if, but there is no else clause.

\[ \text{when } \star \text{debug-flag}\star\text{ print } "\text{you are here}" \]

6.1.2.8 unless

unless test statement

The unless statement is similar to when (and if) but the statement is executed when the test expression is false.

\[ \text{unless count} = 0 \text{ set average} = \text{sum} / \text{count} \]

6.1.2.9 load

load expression

The load command loads a file named by expression, which must evaluate to a string path name for the file. To load a file, SAL interprets each statement in the file, stopping when the end of the file or an error is encountered. If the file ends in .lsp, the file is assumed to contain Lisp expressions, which are evaluated by the XLISP interpreter. In general, SAL files should end with the extension .sal.
6.1.2.10 loop

loop [with-stmt] {stepping}* {stopping}* {action}+ [finally] end

The loop statement is by far the most complex statement in SAL, but it offers great flexibility for just about any kind of iteration. The basic function of a loop is to repeatedly evaluate a sequence of action’s which are statements. Before the loop begins, local variables may be declared in with-stmt, a with statement.

The stepping clauses do several things. They introduce and initialize additional local variables similar to the with-stmt. However, these local variables are updated to new values after the action’s. In addition, some stepping clauses have associated stopping conditions, which are tested on each iteration before evaluating the action’s.

There are also stopping clauses that provide additional tests to stop the iteration. These are also evaluated and tested on each iteration before evaluating the action’s.

When some stepping or stopping condition causes the iteration to stop, the finally clause is evaluated (if present). Local variables and their values can still be accessed in the finally clause. After the finally clause, the loop statement completes.

The stepping clauses are the following:

repeat expression

Sets the number of iterations to the value of expression, which should be an integer (FIXNUM).

for var = expression [ then expr2 ]

Introduces a new local variable named var and initializes it to expression. Before each subsequent iteration, var is set to the value of expr2. If the then part is omitted, expression is re-evaluated and assigned to var on each subsequent iteration. Note that this differs from a with-stmt where expressions are evaluated and variables are only assigned their values once.

for var in expression

Evaluates expression to obtain a list and creates a new local variable initialized to the first element of the list. After each iteration, var is assigned the next element of the list. Iteration stops when var has assumed all values from the list. If the list is initially empty, the loop action’s are not evaluated (there are zero iterations).

for var [from from-expr][[to | below | downto | above] to-expr][by step-expr]

Introduces a new local variable named var and initialized to the value of the expression from-expr (with a default value of 0). After each iteration of the loop, var is incremented by the value of step-expr (with a default value of 1). The iteration ends when var is greater than the value of to-expr if there is a to clause, greater than or equal to the value of to-expr if there is a below clause, less than the value of to-expr if there is a downto clause, or less than or equal to the value of to-expr if there is a above clause. (In the cases of downto and above, the default increment value is -1. If there is no to, below, downto, or above clause, no iteration stop test is created for this stepping clause.

The stopping clauses are the following:
while expression
  The iterations are stopped when expression evaluates to false. Anything not false is considered to mean true.

until expression
  The iterations are stopped when expression evaluates to true.

The finally clause is defined as follows:

finally statement
  The statement is evaluated when one of the stepping or stopping clauses ends the loop. As always, statement may be a begin-end statement. If an action evaluates a return statement, the finally statement is not executed.

Loops often fall into common patterns, such as iterating a fixed number of times, performing an operation on some range of integers, collecting results in a list, and linearly searching for a solution. These forms are illustrated in the examples below.

; iterate 10 times
loop
  repeat 10
  print random(100)
end

; print even numbers from 10 to 20
; note that 20 is printed. On the next iteration,
; i = 22, so i >= 22, so the loop exits.
loop
  for i from 10 to 22 by 2
  print i
end

; collect even numbers in a list
loop
  with lis
  for i from 0 to 10 by 2
    set lis @= i ; push integers on front of list,
    ; which is much faster than append,
    ; but list is built in reverse
  finally set result = reverse(lis)
end
; now, the variable result has a list of evens
; find the first even number in a list
result = #f ; #f means "false"
loop
  for elem in lis
    until evenp(elem)
    finally result = elem
end
; result has first even value in lis (or it is #f)

6.1.2.11 play

play expr
The play statement plays the sound denoted by expr, an expression.

6.1.2.12 plot

plot expr {, dur, n}
The plot statement plots the sound denoted by expr, an expression. If you plot a long sound, the plot statement will by default truncate the sound to 2.0 seconds and resample the signal to 1000 points. The optional dur is an expression that specifies the (maximum) duration to be plotted, and the optional n specifies the number of points to be plotted. Executing a plot statement is equivalent to calling the s-plot function (see Section 7.5).

6.1.2.13 print

print expr {, expr}* The print statement prints the values separated by spaces and followed by a newline. [Note that in the original SAL, the newline is printed before the values, not after.]

6.1.2.14 display

display string {, expression}* The display statement is handy for debugging. At present, it is only implemented in Nyquist SAL. When executed, display prints the string followed by a colon and then, for each expression, the expression and its value are printed; after the last expression, a newline is printed. For example,

display "In function foo", bar, baz

prints
  In function foo : bar = 23, baz = 5.3
Nyquist Reference Manual Chapter 6. SAL

SAL may print the expressions using Lisp syntax, e.g. if the expression is "bar + baz," do not be surprised if the output is "(sum bar baz) = 28.3."

    print "The value of x is", x

6.1.2.15 return

return expression

The return statement can only be used inside a function. It evaluates expression and then the function returns the value of the expression to its caller.

6.1.2.16 set

set var op expression [, var op expression] *

The set statement changes the value of a variable var according to the operator op and the value of the expression. The operators are:

=  The value of expression is assigned to var.

+= The value of expression is added to var.

*=

The value of var is multiplied by the value of the expression.

&=

The value of expression is inserted as the last element of the list referenced by var. If var is the empty list (denoted by #f), then var is assigned a newly constructed list of one element, the value of expression.

^=

The value of expression, a list, is appended to the list referenced by var. If var is the empty list (denoted by #f), then var is assigned the (list) value of expression.

@=

Pushes the value of expression onto the front of the list referenced by var. If var is empty (denoted by #f), then var is assigned a newly constructed list of one element, the value of expression.
Sets the new value of var to the minimum of the old value of var and the value of expression.

Sets the new value of var to the maximum of the old value of var and the value of expression.

; example from Rick Taube's SAL description
loop
  with a, b = 0, c = 1, d = {}, e = {}, f = -1, g = 0
  for i below 5
    set a = i, b += 1, c *= 2, d &= i, e @= i, f <= i, g >= i
  finally display "results", a, b, c, d, e, f, g
end

6.1.2.17 with

with var [= expression] {}, var [= expression]*

The with statement declares and initializes local variables. It can appear only after begin or loop. If the expression is omitted, the initial value is false. The variables are visible only inside the begin-end or loop statement where the with statement appears. Even in loop’s the variables are initialized only when the loop is entered, not on each iteration.

6.1.2.18 exit

exit [nyquist]

The exit statement is unique to Nyquist SAL. It returns from SAL mode to the XLISP interpreter. (Return to SAL mode by typing "(sal)"). If nyquist is included in the statement, then the entire Nyquist process will exit.

6.2 Interoperability of SAL and XLISP

When SAL evaluatas command or loads files, it translates SAL into XLISP. You can think of SAL as a program that translates everything you write into XLISP and entering it for you. Thus, when you define a SAL function, the function actually exists as an XLISP function (created using Lisp’s defun special form). When you set or evaluate global variables in SAL, these are exactly the same Lisp global variables. Thus, XLISP functions can call SAL functions and vice-versa. At run time, everything is Lisp.
6.2.1 Function Calls

In general, there is a very simple translation from SAL to Lisp syntax and back. A function call is SAL, for example,

\[ \text{osc}(g4, 2.0) \]

is translated to Lisp by moving the open parenthesis in front of the function name and removing the commas:

\[ (\text{osc} g4 2.0) \]

Similarly, if you want to translate a Lisp function call to SAL, just reverse the translation.

6.2.2 Symbols and Functions

SAL translates keywords with trailing colons (such as \texttt{foo:}) into Lisp keywords with leading colons (such as \texttt{:foo}), but SAL keywords are not treated as expressions as they are in Lisp. You cannot write \texttt{open("myfile.txt", direction: output:)} because SAL expects an expression after direction. A special form \texttt{keyword} is defined to generate a Lisp keyword as an expression. The argument is the keyword \texttt{without} a colon, e.g. \texttt{open("myfile.txt", direction: \texttt{keyword(output)})}. Alternatively, you can write the Lisp-style keyword with the leading colon, e.g. \texttt{open("myfile.txt", direction: :output)}.

In Nyquist SAL, the hash character (#), can be used as a prefix to a Lisp function name. For example, the following command is not legal because \texttt{print} is a SAL command name, not a legal function name:

\[ \text{set } v = \text{append}(\text{print(a)}, \text{print(b)}) \]

. (Here the intent is to print arguments to append). However, you can use the hash character to access the Lisp \texttt{print} function:

\[ \text{set } v = \text{append}(\#\text{print(a)}, \#\text{print(b)}) \]

6.2.3 Playing Tricks On the SAL Compiler

In many cases, the close coupling between SAL and XLISP gives SAL unexpected expressive power. A good example is \texttt{seqrep}. This is a special looping construct in Nyquist, implemented as a macro in XLISP. In Lisp, you would write something like:

\[ (\text{seqrep}(i 10) \ (\text{pluck c4})) \]

One might expect SAL would have to define a special \texttt{seqrep} statement to express this, but since statements do not return values, this approach would be problematic. The solution (which is already fully implemented in Nyquist) is to define a new macro \texttt{sal-seqrep} that is equivalent to \texttt{seqrep} except that it is called as follows:

\[ (\text{sal-seqrep} i 10 \ (\text{pluck c4})) \]
The SAL compiler automatically translates the identifier `seqrep` to `sal-seqrep`. Now, in SAL, you can just write

```sal
seqrep(i, 10, pluck(c4))
```

which is translated in a pretty much semantics-unaware fashion to

```sal
(sal-seqrep i 10 (pluck c4))
```

and viola!, we have Nyquist control constructs in SAL even though SAL is completely unaware that `seqrep` is actually a special form.
7 Nyquist Functions

This chapter provides a language reference for Nyquist. Operations are categorized by functionality and abstraction level. Nyquist is implemented in two important levels: the "high level" supports behavioral abstraction, which means that operations like `stretch` and `at` can be applied. These functions are the ones that typical users are expected to use, and most of these functions are written in XLISP.

The "low-level" primitives directly operate on sounds, but know nothing of environmental variables (such as `*warp*`, etc.). The names of most of these low-level functions start with "snd-". In general, programmers should avoid any function with the "snd-" prefix. Instead, use the "high-level" functions, which know about the environment and react appropriately. The names of high-level functions do not have prefixes like the low-level functions.

There are certain low-level operations that apply directly to sounds (as opposed to behaviors) and are relatively "safe" for ordinary use. These are marked as such.

Nyquist uses both linear frequency and equal-temperament pitch numbers to specify repetition rates. Frequency is always specified in either cycles per second (hz), or pitch numbers, also referred to as "steps," as in steps of the chromatic scale. Steps are floating point numbers such that 60 = Middle C, 61 = C#, 61.23 is C# plus 23 cents, etc. The mapping from pitch number to frequency is the standard exponential conversion, and fractional pitch numbers are allowed:

\[
\text{frequency} = 440 \times 2^{(\text{pitch} - 69)/12}
\]

There are many predefined pitch names. By default these are tuned in equal temperament, with A4 = 440Hz, but these may be changed. (See Section 1.8).

7.1 Sounds

A sound is a primitive data type in Nyquist. Sounds can be created, passed as parameters, garbage collected, printed, and set to variables just like strings, atoms, numbers, and other data types.

7.1.1 What is a Sound?

Sounds have 5 components:
• **srate** – the sample rate of the sound.

• **samples** – the samples.

• **signal-start** – the time of the first sample.

• **signal-stop** – the time of one past the last sample.

• **logical-stop** – the time at which the sound logically ends, e.g. a sound may end at the beginning of a decay. This value defaults to **signal-stop**, but may be set to any value.

It may seem that there should be **logical-start** to indicate the logical or perceptual beginning of a sound as well as a **logical-stop** to indicate the logical ending of a sound. In practice, only **logical-stop** is needed; this attribute tells when the next sound should begin to form a sequence of sounds. In this respect, Nyquist sounds are asymmetric: it is possible to compute sequences forward in time by aligning the logical start of each sound with the **logical-stop** of the previous one, but one cannot compute "backwards", aligning the logical end of each sound with the logical start of its successor. The root of this asymmetry is the fact that when we invoke a behavior, we say when to start, and the result of the behavior tells us its logical duration. There is no way to invoke a behavior with a direct specification of when to stop\(^1\).

*Note:* there is no way to enforce the intended "perceptual" interpretation of **logical-stop**. As far as Nyquist is concerned, these are just numbers to guide the alignment of sounds within various control constructs.

### 7.1.2 Multichannel Sounds

Multichannel sounds are represented by Lisp arrays of sounds. To create an array of sounds the XLISP vector function is useful. Most low-level Nyquist functions (the ones starting with `snd-`) do not operate on multichannel sounds. Most high-level functions do operate on multichannel sounds.

### 7.1.3 Accessing and Creating Sound

Several functions display information concerning a sound and can be used to query the components of a sound. There are functions that access samples in a sound and functions that construct sounds from samples.

\[
\text{sref}(\text{sound}, \text{time}) \quad \text{[SAL]}
\]

\[
(s\text{ref sound time}) \quad \text{[LISP]}
\]

Accesses **sound** at the point **time**, which is a local time. If **time** does not correspond to a sample time, then the nearest samples are linearly interpolated to form the result. To access a particular sample, either convert the sound to an array (see `snd-samples` below), or use `snd-srate` and `snd-t0` (see below) to find the sample rate and starting time, and compute a time (\(t\)) from the sample number (\(n\)):

\(^1\)Most behaviors will stop at time 1, warped according to `*warp*` to some real time, but this is by convention and is not a direct specification.
\[ t = \left( \frac{n}{srate} \right) + t0 \]

Thus, the Lisp code to access the \( n \)th sample of a sound would look like:

\[
\text{(sref sound (global-to-local (+ (/ n (snd-srate sound)) (snd-t0 sound))))}
\]

Or in SAL, it would look like:

\[
sref(sound, \text{global-to-local}(n / \text{snd-srate}(sound) + \text{snd-t0}(sound)))
\]

Here is why \text{sref} interprets its time argument as a local time (shown first in LISP and then in SAL syntax):

\[
\quad > \text{(sref (ramp 1) 0.5)}; \text{evaluate a ramp at time 0.5}
\]

\[
0.5
\]

\[
\quad \text{SAL> print sref(ramp(1), 0.5); evaluate a ramp at time 0.5}
\]

\[
0.5
\]

\[
\quad > (\text{at 2.0 (sref (ramp 1) 0.5)}); \text{ramp is shifted to start at 2.0}
\]

\[
; \text{the time, 0.5, is shifted to 2.5}
\]

\[
0.5
\]

\[
\quad \text{SAL> sref(ramp(1), 0.5) @ 2.0; ramp is shifted to start at 2.0}
\]

\[
; \text{the time, 0.5, is shifted to 2.5}
\]

\[
0.5
\]

If you were to use \text{snd-sref}, which treats time as global, instead of \text{sref}, which treats time as local, then the first example above would return the same answer (0.5), but the second example would return 0. Why? Because the ramp behavior would be shifted to start at time 2.0, but the resulting sound would be evaluated at global time 0.5. By definition, sounds have a value of zero before their start time.

\[
\text{sref-inverse(sound, value)} \ [\text{SAL}]
\]

\[
\text{(sref-inverse sound value)} \ [\text{LISP}]
\]

Search \text{sound} for the first point at which it achieves \text{value} and return the corresponding (linearly interpolated) time. If no inverse exists, an error is raised. This function is used by Nyquist in the implementation of time warping.

\[
\text{snd-from-array(t0, sr, array)} \ [\text{SAL}]
\]

\[
\text{(snd-from-array t0 sr array)} \ [\text{LISP}]
\]

Converts a lisp array of \text{FLONUM}s into a sound with starting time \text{t0} and sample rate \text{sr}. Safe for ordinary use. Be aware that arrays of floating-point samples use 14 bytes per sample, and an additional 4 bytes per sample are allocated by this function to create a sound type.

\[
\text{snd-fromarraystream(t0, sr, object)} \ [\text{SAL}]
\]

\[
\text{(snd-fromarraystream t0 sr object)} \ [\text{LISP}]
\]

Creates a sound for which samples come from \text{object}. The starting time is \text{t0} (a \text{FLONUM}), and the sample rate is \text{sr}. The \text{object} is an XLISP object (see Section C.11 for information on objects.) A sound is returned. When the sound needs samples, they are generated by sending the message :next to \text{object}. If \text{object} returns NIL, the sound terminates. Otherwise, \text{object} must return an array of \text{FLONUM}s. The values in these arrays are concatenated to form the samples of the resulting sound. There is no provision for \text{object} to specify the logical stop time of the sound, so the logical stop time is the termination time.
\textbf{snd-fromobject}(t0, sr, object) [SAL]

\texttt{(snd-fromobject t0 sr object) [LISP]}

Creates a sound for which samples come from \textit{object}. The starting time is \textit{t0} (a \texttt{FLONUM}), and the sample rate is \textit{sr}. The \textit{object} is an XLISP object (see Section C.11 for information on objects. A sound is returned. When the sound needs samples, they are generated by sending the message :\texttt{next} to \textit{object}. If \textit{object} returns \texttt{NIL}, the sound terminates. Otherwise, \textit{object} must return a \texttt{FLONUM}. There is no provision for \textit{object} to specify the logical stop time of the sound, so the logical stop time is the termination time.

\textbf{snd-extent}(sound, maxsamples) [SAL]

\texttt{(snd-extent sound maxsamples) [LISP]}

Returns a list of two numbers: the starting time of \textit{sound} and the terminate time of \textit{sound}. Finding the terminate time requires that samples be computed. Like most Nyquist functions, this is non-destructive, so memory will be allocated to preserve the sound samples. If the sound is very long or infinite, this may exhaust all memory, so the \textit{maxsamples} parameter specifies a limit on how many samples to compute. If this limit is reached, the terminate time will be (incorrectly) based on the sound having \textit{maxsamples} samples. This function is safe for ordinary use.

\textbf{snd-fetch}(sound) [SAL]

\texttt{(snd-fetch sound) [LISP]}

Reads samples sequentially from \textit{sound}. This returns a \texttt{FLONUM} after each call, or \texttt{NIL} when \textit{sound} terminates. \textbf{Note}: \texttt{snd-fetch} modifies \textit{sound}; it is strongly recommended to copy \textit{sound} using \texttt{snd-copy} and access only the copy with \texttt{snd-fetch}.

\textbf{snd-fetch-array}(sound, len, step) [SAL]

\texttt{(snd-fetch-array sound len step) [LISP]}

Reads sequential arrays of samples from \textit{sound}, returning either an array of \texttt{FLONUM}s or \texttt{NIL} when \textit{sound} terminates. The \textit{len} parameter, a \texttt{FIXNUM}, indicates how many samples should be returned in the result array. After the array is returned, \textit{sound} is modified by skipping over \textit{step} (a \texttt{FIXNUM}) samples. If \textit{step} equals \textit{len}, then every sample is returned once. If \textit{step} is less than \textit{len}, each returned array will overlap the previous one, so some samples will be returned more than once. If \textit{step} is greater than \textit{len}, then some samples will be skipped and not returned in any array. The \textit{step} and \textit{len} may change at each call, but in the current implementation, an internal buffer is allocated for \textit{sound} on the first call, so subsequent calls may not specify a greater \textit{len} than the first. When an array is returned, it will have \textit{len} samples. If necessary, \texttt{snd-fetch-array} will read zeros beyond the end of the sound to fill the array. When this happens, *\texttt{rslt}* is set to a \texttt{FIXNUM} number of samples in the array that were read from the sound before the physical stop time of the sound. If all samples in the array are "valid" samples from the sound (coming from the sound before the sound terminates), *\texttt{rslt}* is set to \texttt{NIL}. The *\texttt{rslt}* variable is global and used to return extra results from other functions, so programs should not assume *\texttt{rslt}* is valid after subsequent function calls. \textbf{Note}: \texttt{snd-fetch-array} modifies \textit{sound}; it is strongly recommended to copy \textit{sound} using \texttt{snd-copy} and access only the copy with \texttt{snd-fetch-array}. 

v 3.12

69
snd-flatten(sound, maxlen) [SAL]
(snd-flatten sound maxlen) [LISP]

This function is identical to snd-length. You would use this function to force samples to be computed in memory. Normally, this is not a good thing to do, but here is one appropriate use: In the case of sounds intended for wavetables, the unevaluated sound may be larger than the evaluated (and typically short) one. Calling snd-flatten will compute the samples and allow the unit generators to be freed in the next garbage collection. Note: If a sound is computed from many instances of table-lookup oscillators, calling snd-flatten will free the oscillators and their tables. Calling (stats) will print how many total bytes have been allocated to tables.

snd-length(sound, maxlen) [SAL]
(snd-length sound maxlen) [LISP]

Counts the number of samples in sound up to the physical stop time. If the sound has more than maxlen samples, maxlen is returned. Calling this function will cause all samples of the sound to be computed and saved in memory (about 4 bytes per sample). Otherwise, this function is safe for ordinary use.

snd-maxsamp(sound) [SAL]
(snd-maxsamp sound) [LISP]

Computes the maximum of the absolute value of the samples in sound. Calling this function will cause samples to be computed and saved in memory. (This function should have a maxlen parameter to allow self-defense against sounds that would exhaust available memory.) Otherwise, this function is safe for ordinary use. This function will probably be removed in a future version. See peak, a replacement (page 117).

snd-play(expression) [SAL]
(snd-play expression) [LISP]

Evaluates expression to obtain a sound or array of sounds, computes all of the samples (without retaining them in memory), and returns. Originally, this was a placeholder for a facility to play samples directly to an audio output device, but playback is now accomplished by s-save. Meanwhile, since this function does not save samples in memory or write them to a disk, it is useful in determining how much time is spent calculating samples. See s-save (Section 7.5) for saving samples to a file, and play (Section 7.5) to play a sound. This function is safe for ordinary use.

snd-print-tree(sound) [SAL]
(snd-print-tree sound) [LISP]

Prints an ascii representation of the internal data structures representing a sound. This is useful for debugging Nyquist. This function is safe for ordinary use.

snd-samples(sound, limit) [SAL]
(snd-samples sound limit) [LISP]

Converts the samples into a lisp array. The data is taken directly from the samples, ignoring shifts. For example, if the sound starts at 3.0 seconds, the first sample will refer to time 3.0, not time 0.0. A maximum of limit samples is returned. This function is safe for ordinary use, but like snd-from-array, it requires a total of slightly over 18 bytes per sample.
Nyquist Reference Manual

Chapter 7. Nyquist Functions

snd-srate(sound) [SAL]
(snd-srate sound) [LISP]

Returns the sample rate of the sound. Safe for ordinary use.

snd-time(sound) [SAL]
(snd-time sound) [LISP]

Returns the start time of the sound. This will probably go away in a future version, so use snd-t0 instead.

snd-t0(sound) [SAL]
(snd-t0 sound) [LISP]

Returns the time of the first sample of the sound. Note that Nyquist operators such as add always copy the sound and are allowed to shift the copy up to one half sample period in either direction to align the samples of two operands. Safe for ordinary use.

snd-print(expression, maxlen) [SAL]
(snd-print expression maxlen) [LISP]

Evaluates expression to yield a sound or an array of sounds, then prints up to maxlen samples to the screen (stdout). This is similar to snd-save, but samples appear in text on the screen instead of in binary in a file. This function is intended for debugging. Safe for ordinary use.

snd-set-logical-stop(sound, time) [SAL]
(snd-set-logical-stop sound time) [LISP]

Returns a sound which is sound, except that the logical stop of the sound occurs at time. Note: do not call this function. When defining a behavior, use set-logical-stop or set-logical-stop-abs instead.

snd-sref(sound, time) [SAL]
(snd-sref sound time) [LISP]

Evaluates sound at the global time given by time. Safe for ordinary use, but normally, you should call sref instead.

snd-stop-time(sound) [SAL]
(snd-stop-time sound) [LISP]

Returns the stop time of sound. Sounds can be "clipped" or truncated at a particular time. This function returns that time or MAX-STOP-TIME if the programmer has not specified a stop time for the sound. Safe for ordinary use.

soundp(sound) [SAL]
(soundp sound) [LISP]

Returns true iff sound is a SOUND. Safe for ordinary use.

stats() [SAL]
(stats) [LISP]

Prints the memory usage status. See also the XLISP mem function. Safe for ordinary use. This is the only way to find out how much memory is being used by table-lookup oscillator instances.
Nyquist Reference Manual

Chapter 7. Nyquist Functions

snd-set-max-audio-mem(bytes) [SAL]
(snd-set-max-audio-mem bytes) [LISP]
Sets a limit (a FIXNUM in bytes) on the amount of main memory that Nyquist will allocate for
SOUNDs (this does not count table memory). The default is about 1GB. The return value is the
previous limit, in bytes. This is not a limit on how big sounds can be. Since Nyquist computes sound
incrementally, it can generally play sounds or write sounds to files without storing them in memory.
However, if you store large sounds in variables, memory usage can exceed your available RAM, causing
extremely slow computation as main memory is swapped to and from disk or Nyquist might run out of
memory and crash. The goal of placing a limit on audio memory is to terminate computations before
memory is totally exhausted, allowing Nyquist to print an error message and allowing the user to view
or save work. Generally, if you see the message "The maximum number of sample blocks has been
reached, ...." you should fix your code to avoid accumulating samples in memory, e.g. do not assign
sounds to global variables. Alternatively, you can use this function to increase the limit, but of course
you are still limited by the actual size of memory on your computer, and exceeding that could cause
severe performance problems in Nyquist and any other applications that are running. See also
Section 1.4 for command line options to limit yquist memory, run time, and file access (these options
are not available through the NyquistIDE.)

7.1.4 Miscellaneous Functions

These are all safe and recommended for ordinary use.

db-to-linear(x) [SAL]
(db-to-linear x) [LISP]
Returns the conversion of x from decibels to linear. 0dB is converted to 1. 20dB represents a linear
factor of 10. If x is a sound, each sample is converted and a sound is returned. If x is a multichannel
sound, each channel is converted and a multichannel sound (array) is returned. Note: With sounds,
conversion is only performed on actual samples, not on the implicit zeros before the beginning and
after the termination of the sound. Sample rates, start times, etc. are taken from x

db-to-vel(x [, float]) [SAL]
(db-to-vel x [float]) [LISP]
Returns the conversion of x from decibels to MIDI velocity using a rule that maps -60 dB to 1 and 0 dB
to 127. The MIDI velocity varies linearly with the square root of amplitude. The default value of float
is nil and the result is a FIXNUM clipped to fall in the legal range of 1-127, but if a non-nil value is
provided, the result is a FLONUM that is not rounded or clipped. The input parameter must be a FIXNUM
or FLONUM. Sounds are not allowed.

follow(sound, floor, risetime, falltime, lookahead) [SAL]
(follow sound floor risetime falltime lookahead) [LISP]
An envelope follower intended as a component for compressor and limiter functions. The basic goal
of this function is to generate a smooth signal that rides on the peaks of the input signal. The usual
objective is to produce an amplitude envelope given a low-sample rate (control rate) signal representing local RMS measurements. The first argument is the input signal. The floor is the minimum output value. The risetime is the time (in seconds) it takes for the output to rise (exponentially) from floor to unity (1.0) and the falltime is the time it takes for the output to fall (exponentially) from unity to floor. The algorithm looks ahead for peaks and will begin to increase the output signal according to risetime in anticipation of a peak. The amount of anticipation (in seconds) is given by lookahead. The algorithm is as follows: the output value is allowed to increase according to risetime or decrease according to falltime. If the next input sample is in this range, that sample is simply output as the next output sample. If the next input sample is too large, the algorithm goes back in time as far as necessary to compute an envelope that rises according to risetime to meet the new value. The algorithm will only work backward as far as lookahead. If that is not far enough, then there is a final forward pass computing a rising signal from the earliest output sample. In this case, the output signal will be at least momentarily less than the input signal and will continue to rise exponentially until it intersects the input signal. If the input signal falls faster than indicated by falltime, the output fall rate will be limited by falltime, and the fall in output will stop when the output reaches floor. This algorithm can make two passes through the buffer on sharply rising inputs, so it is not particularly fast. With short buffers and low sample rates this should not matter. See snd-avg for a function that can help to generate a low-sample-rate input for follow. See snd-chase in Section 7.6.3 for a related filter.

\texttt{gate(sound, lookahead, risetime, falltime, floor, threshold)} [SAL]
\texttt{(gate sound lookahead risetime falltime floor threshold)} [LISP]
Generate an exponential rise and decay intended for noise gate implementation. The decay starts when the signal drops below threshold and stays there for longer than lookahead (a FLONUM in seconds). (The signal begins to drop when the signal crosses threshold, not after lookahead.) Decay continues until the value reaches floor (a FLONUM), at which point the decay stops and the output value is held constant. Either during the decay or after the floor is reached, if the signal goes above threshold, then the output value will rise to unity (1.0) at the point the signal crosses the threshold. Because of internal lookahead, the signal actually begins to rise before the signal crosses threshold. The rise is a constant-rate exponential and set so that a rise from floor to unity occurs in risetime. Similary, the fall is a constant-rate exponential such that a fall from unity to floor takes falltime.

\texttt{noise-gate(sound [lookahead, risetime, falltime, floor, threshold])} [SAL]
\texttt{(noise-gate sound [lookahead risetime falltime floor threshold])} [LISP]
A simple noise gate implementation based on gate. All parameters except snd are optional and default values are lookahead: 0.5, risetime: 0.02, falltime: 0.5, floor: 0.01, threshold: 0.01. The result is the input snd, where below-threshold segments of sound are silenced.

\texttt{hz-to-step(freq)} [SAL]
\texttt{(hz-to-step freq)} [LISP]
Returns a step number for freq (in hz), which can be either a number of a SOUND. The result has the same type as the argument. See also \texttt{step-to-hz} (below).

\texttt{linear-to-db(x)} [SAL]
\texttt{(linear-to-db x)} [LISP]
Returns the conversion of x from linear to decibels. 1 is converted to 0. 0 is converted to -INF (a special
IEEE floating point value.) A factor of 10 represents a 20dB change. If \( x \) is a sound, each sample is converted and a sound is returned. If \( x \) is a multichannel sound, each channel is converted and a multichannel sound (array) is returned. **Note:** With sounds, conversion is only performed on actual samples, not on the implicit zeros before the beginning and after the termination of the sound. Start times, sample rates, etc. are taken from \( x \).

\[
\text{linear-to-vel}(x [, \text{float}]) \quad \text{[SAL]}
\]
\[
\text{(linear-to-vel \( x \) [\text{float}])} \quad \text{[LISP]}
\]

Returns the conversion of \( x \) from linear amplitude to MIDI velocity using a rule that maps -60 dB to 1 and 0 dB to 127. The MIDI velocity varies linearly with the square root of amplitude. The default value of \( \text{float} \) is \text{nil} and the result is a \text{FIXNUM} clipped to fall in the legal range of 1-127, but if a non-nil value is provided, the result is a \text{FLONUM} that is not rounded or clipped. The input parameter must be a \text{FIXNUM} or \text{FLONUM}. Sounds are not allowed.

\[
\log(x) \quad \text{[SAL]}
\]
\[
(\log x) \quad \text{[LISP]}
\]

Calculates the natural log of \( x \) (a \text{FLONUM}). (See \text{s-log} for a version that operates on signals.)

\[
\text{set-control-srate}(\text{rate}) \quad \text{[SAL]}
\]
\[
(\text{set-control-srate rate}) \quad \text{[LISP]}
\]

Sets the default sampling rate for control signals to \( \text{rate} \) by setting *default-control-srate* and reinitializing the environment. Do not call this within any synthesis function (see the control-srate-abs transformation, Section 7.3).

\[
\text{set-sound-srate}(\text{rate}) \quad \text{[SAL]}
\]
\[
(\text{set-sound-srate rate}) \quad \text{[LISP]}
\]

Sets the default sampling rate for audio signals to \( \text{rate} \) by setting *default-sound-srate* and reinitializing the environment. Do not call this within any synthesis function (see the sound-srate-abs transformation, Section 7.3).

\[
\text{set-pitch-names}() \quad \text{[SAL]}
\]
\[
(\text{set-pitch-names}) \quad \text{[LIS]}
\]

Initializes pitch variables (c0, cs0, df0, d0, ... b0, c1, ... b7). A440 (the default tuning) is represented by the step 69.0, so the variable a4 (fourth octave A) is set to 69.0. You can change the tuning by setting *A4-Hertz* to a value (in Hertz) and calling set-pitch-names to reinitialize the pitch variables. Note that this will result in non-integer step values. It does not alter the mapping from step values to frequency. There is no built-in provision for stretched scales or non-equal temperament, although users can write or compute any desired fractional step values.

\[
\text{step-to-hz}(\text{pitch}) \quad \text{[SAL]}
\]
\[
(\text{step-to-hz pitch}) \quad \text{[LISP]}
\]

Returns a frequency in hz for \( \text{pitch} \), a step number or a \text{SOUND} type representing a time-varying step number. The result is a \text{FLONUM} if \( \text{pitch} \) is a number, and a \text{SOUND} if \( \text{pitch} \) is a \text{SOUND}. See also hz-to-step (above).
get-duration \( \text{dur} \) [SAL]
\begin{align*}
\text{(get-duration dur)} \quad \text{[LISP]}
\end{align*}

Gets the actual duration of something starting at a local time of 0 and ending at a local time of \( \text{dur} \) times the current sustain. For convenience, \(*\text{rslt}*\) is set to the global time corresponding to local time zero.

get-loud() [SAL]
\begin{align*}
\text{(get-loud)} \quad \text{[LISP]}
\end{align*}

Gets the current value of the \(*\text{loud}*\) environment variable. If \(*\text{loud}*\) is a signal, it is evaluated at local time 0 and a number (FLONUM) is returned.

get-sustain() [SAL]
\begin{align*}
\text{(get-sustain)} \quad \text{[LISP]}
\end{align*}

Gets the current value of the \(*\text{sustain}*\) environment variable. If \(*\text{sustain}*\) is a signal, it is evaluated at local time 0 and a number (FLONUM) is returned.

get-transpose() [SAL]
\begin{align*}
\text{(get-transpose)} \quad \text{[LISP]}
\end{align*}

Gets the current value of the \(*\text{transpose}*\) environment variable. If \(*\text{transpose}*\) is a signal, it is evaluated at local time 0 and a number (FLONUM) is returned.

get-warp() [SAL]
\begin{align*}
\text{(get-warp)} \quad \text{[LISP]}
\end{align*}

Gets a function corresponding to the current value of the \(*\text{warp}*\) environment variable. For efficiency, \(*\text{warp}*\) is stored in three parts representing a shift, a scale factor, and a continuous warp function. \text{Get-warp} is used to retrieve a signal that maps logical time to real time. This signal combines the information of all three components of \(*\text{warp}*\) into a single signal. If the continuous warp function component is not present (indicating that the time warp is a simple combination of \textit{at} and \textit{stretch} transformations), an error is raised. This function is mainly for internal system use. In the future, \text{get-warp} will probably be reimplemented to always return a signal and never raise an error.

local-to-global \( \text{local-time} \) [SAL]
\begin{align*}
\text{(local-to-global local-time)} \quad \text{[LISP]}
\end{align*}

Converts a score (local) time to a real (global) time according to the current environment.

round \( x \) [SAL]
\begin{align*}
\text{(round x)} \quad \text{[LISP]}
\end{align*}

Round \( x \) to the nearest integer. If \( x = n + 0.5 \), where \( n \) is an integer, then return \( n + 1 \), even if \( n \) is negative.

snd-set-latency \( \text{latency} \) [SAL]
\begin{align*}
\text{(snd-set-latency latency)} \quad \text{[LISP]}
\end{align*}

Set the latency requested when Nyquist plays sound to \textit{latency}, a FLONUM. The previous value is returned. The default is 0.3 seconds. To avoid glitches, the latency should be greater than the time required for garbage collection and message printing and any other system activity external to Nyquist.
vel-to-db(x) [SAL]  
(vel-to-db x) [LISP]  
Returns the conversion of x from MIDI velocity to decibels using a rule that maps MIDI velocity 1 to
-60 dB and 127 to 0 dB. The amplitude is proportional to the square of MIDI velocity. The input x can
be a FIXNUM or FLONUM but not a sound. The result is a FLONUM.

vel-to-linear(x) [SAL]  
(vel-to-linear x) [LISP]  
Returns the conversion of x from MIDI velocity to linear amplitude ratio using a rule that maps MIDI
velocity 1 to -60 dB (0.001) and 127 to unity gain. The amplitude is proportional to the square of MIDI
velocity. The input x can be a FIXNUM or FLONUM but not a sound. The result is a FLONUM.

7.2 Behaviors

7.2.1 Using Previously Created Sounds

These behaviors take a sound and transform that sound according to the environment. These are useful when
writing code to make a high-level function from a low-level function, or when cuing sounds which were
previously created:

cue(sound) [SAL]  
(cue sound) [LISP]  
Applies *loud*, the starting time from *warp*, *start*, and *stop* to sound.

cue-file(filename) [SAL]  
(cue-file filename) [LISP]  
Same as cue, except the sound comes from the named file, samples from which are coerced to the
current default *sound-srate* sample rate.

sound(sound) [SAL]  
(sound sound) [LISP]  
Applies *loud*, *warp*, *start*, and *stop* to sound.

control(sound) [SAL]  
(control sound) [LISP]  
This function is identical to sound, but by convention is used when sound is a control signal rather
than an audio signal.

7.2.2 Sound Synthesis

These functions provide musically interesting creation behaviors that react to their environment; these are the
"unit generators" of Nyquist:
const(value [, duration]) [SAL]
(const value [duration]) [LISP]
  Creates a constant function at the *control-srate*. Every sample has the given value, and the default duration is 1.0. See also s-rest, which is equivalent to calling const with zero, and note that you can pass scalar constants (numbers) to sim, sum, and mult where they are handled more efficiently than constant functions.

env(t_1, t_2, t_4, l_1, l_2, l_3, [dur]) [SAL]
(env t_1 t_2 t_4 l_1 l_2 l_3 dur) [LISP]
  Creates a 4-phase envelope. t_i is the duration of phase i, and l_i is the final level of phase i. t_3 is implied by the duration dur, and l_4 is 0.0. If dur is not supplied, then 1.0 is assumed. The envelope duration is the product of dur, *stretch*, and *sustain*. If t_1 + t_2 + 2ms + t_4 is greater than the envelope duration, then a two-phase envelope is substituted that has an attack/release time ratio of t_1/t_4. The sample rate of the returned sound is *control-srate*. (See pwl for a more general piece-wise linear function generator.) The effect of time warping is to warp the starting time and ending time. The intermediate breakpoints are then computed as described above.

exp-dec(hold, halfdec, length) [SAL]
(exp-dec hold halfdec length) [LISP]
  This convenient envelope shape is a special case of pwev (see Section 7.2.2.2). The envelope starts at 1 and is constant for hold seconds. It then decays with a half life of halfdec seconds until length. (The total duration is length.) In other words, the amplitude falls by half each halfdec seconds. When stretched, this envelope scales linearly, which means the hold time increases and the half decay time increases.

force-srate(srate, sound) [SAL]
(force-srate srate sound) [LISP]
  Returns a sound which is up- or down-sampled to srate. Interpolation is linear, and no prefiltering is applied in the down-sample case, so aliasing may occur. See also resample.

lfo(freq [, duration, table, phase]) [SAL]
(lfo freq duration table phase) [LISP]
  Just like osc (below) except this computes at the *control-srate* and frequency is specified in Hz. Initial phase is specified in degrees, defaulting to zero. The *transpose* and *sustain* is not applied. The effect of time warping is to warp the starting and ending times. The signal itself will have a constant unwarped frequency.

fmlfo(freq [, table, phase]) [SAL]
(fmlfo freq [table phase]) [LISP]
  A low-frequency oscillator that computes at the *control-srate* using a sound to specify a time-varying frequency in Hz. Initial phase is a FLONUM in degrees. The duration of the result is determined by freq.

maketable(sound) [SAL]
(maketable sound) [LISP]
  Assumes that the samples in sound constitute one period of a wavetable, and returns a wavetable
suitable for use as the table argument to the osc function (see below). Currently, tables are limited to 1,000,000 samples. This limit is the compile-time constant max_table_len set in sound.h. A wavetable is a list of the form

\[(\text{sound pitch-number periodic})\]

where the first element is a sound, the second is the pitch of the sound (this is not redundant, because the sound may represent any number of periods), and the third element is T if the sound is one period of a periodic signal, or nil if the sound is a sample that should not be looped. Wavetables are used by osc, osc, hzosc, amosc, fmosc, lfo, and fmlfo.

**build-harmonic**

(build-harmonic \(n\) \(table-size\) ) [SAL]

\((\text{build-harmonic } n \text{ table-size} )\) [LISP]

Intended for constructing wavetables, this function returns a sound of length \(table-size\) samples containing \(n\) periods of a sinusoid. These can be scaled and summed to form a waveform with the desired harmonic content. See page 14 for an example. A scaled sum of these harmonics can be passed to maketable to construct a wavetable suitable for osc and other oscillators.

**control-warp**

(control-warp \(warp\)-fn \(signal\) [ \(wrate\)] ) [SAL]

\((\text{control-warp } warp\text{-fn } signal \text{ wrate})\) [LISP]

Applies a warp function \(warp\text{-fn}\) to \(signal\) using function composition. If \(wrate\) is omitted, linear interpolation is used. \(warp\text{-fn}\) is a mapping from score (logical) time to real time, and \(signal\) is a function from score time to real values. The result is a function from real time to real values at a sample rate of \(*\text{control-srate}*\). See sound-warp for an explanation of \(wrate\) and high-quality warping.

**mult**

\(\text{mult}(beh_1, beh_2, ... )\) [SAL]

\((\text{mult } beh_1 \text{ beh}_2 ... )\) [LISP]

Returns the product of behaviors. The arguments may also be numbers, in which case simple multiplication is performed. If a number and sound are mixed, the scale function is used to scale the sound by the number. When sounds are multiplied, the resulting sample rate is the maximum sample rate of the factors.

**prod**

\(\text{prod}(beh_1, beh_2, ... )\) [SAL]

\((\text{prod } beh_1 \text{ beh}_2 ... )\) [LISP]

Same as mult.

**pan**

\(\text{pan}(sound, where)\) [SAL]

\((\text{pan } sound \text{ where})\) [LISP]

Pans \(sound\) (a behavior) according to \(where\) (another behavior or a number). Sound must be monophonic. Where may be a monophonic sound (e.g. (ramp)) or simply a number (e.g. 0.5). In either case, \(where\) should range from 0 to 1, where 0 means pan completely left, and 1 means pan completely right. For intermediate values, the sound to each channel is scaled linearly. Presently, pan does not check its arguments carefully.

**prod**

\(\text{prod}(beh_1, beh_2, ... )\) [SAL]

\((\text{prod } beh_1 \text{ beh}_2 ... )\) [LISP]

Same as mult.
Figure 7.1: The `shift-time` function shifts a sound in time according to its `shift` argument.

```lisp
(resample sound srate) [SAL]
(resample sound srate) [LISP]
Similar to `force-srate`, except high-quality interpolation is used to prefilter and reconstruct the signal at the new sample rate. Also, the result is scaled by 0.95 to reduce problems with clipping. (See also `sound-warp`.)

(scale scale sound) [SAL]
(scale scale sound) [LISP]
Scales the amplitude of `sound` by the factor `scale`. Identical function to `snd-scale`, except that it handles multichannel sounds. Sample rates, start times, etc. are taken from `sound`.

(scale-db db sound) [SAL]
(scale-db db sound) [LISP]
Scales the amplitude of `sound` by the factor `db`, expressed in decibels. Sample rates, start times, etc. are taken from `sound`.

(scale-srate sound scale) [SAL]
(scale-srate sound scale) [LISP]
Scales the sample rate of `sound` by `scale` factor. This has the effect of linearly shrinking or stretching time (the sound is not upsampled or downsampled). This is a special case of `snd-xform` (see Section 7.6.2).

(shift-time sound shift) [SAL]
(shift-time sound shift) [LISP]
Shift `sound` by `shift` seconds. If the sound is

See Figure 7.1. This is a special case of `snd-xform` (see Section 7.6.2).
sound-warp(\textit{warp-fn}, \textit{signal} [, \textit{wrate}]) [SAL]
(sound-warp \textit{warp-fn} \textit{signal} [\textit{wrate}]) [LISP]
Applies a warp function \textit{warp-fn} to \textit{signal} using function composition. If the optional parameter \textit{wrate}
is omitted or NIL, linear interpolation is used. Otherwise, high-quality sample interpolation is used, and the result is scaled by 0.95 to reduce problems with clipping (interpolated samples can exceed the peak values of the input samples.) \textit{warp-fn} is a mapping from score (logical) time to real time, and \textit{signal} is a function from score time to real values. The result is a function from real time to real values at a sample rate of *\textit{sound-srate}*. See also \texttt{control-warp}.

If \textit{wrate} is not NIL, it must be a number. The parameter indicates that high-quality resampling should be used and specifies the sample rate for the inverse of \textit{warp-fn}. Use the lowest number you can. (See below for details.) Note that high-quality resampling is much slower than linear interpolation.

To perform high-quality resampling by a fixed ratio, as opposed to a variable ratio allowed in \texttt{sound-warp}, use \texttt{scale-srate} to stretch or shrink the sound, and then \texttt{resample} to restore the original sample rate.

\texttt{Sound-warp} and \texttt{control-warp} both take the inverse of \textit{warp-fn} to get a function from real time to score time. Each sample of this inverse is thus a score time; \textit{signal} is evaluated at each of these score times to yield a value, which is the desired result. The sample rate of the inverse warp function is somewhat arbitrary. With linear interpolation, the inverse warp function sample rate is taken to be the output sample rate. Note, however, that the samples of the inverse warp function are stored as 32-bit floats, so they have limited precision. Since these floats represent sample times, rounding can be a problem. Rounding in this case is equivalent to adding jitter to the sample times. Nyquist ignores this problem for ordinary warping, but for high-quality warping, the jitter cannot be ignored.

The solution is to use a rather low sample rate for the inverse warp function. \texttt{Sound-warp} can then linearly interpolate this signal using double-precision floats to minimize jitter between samples. The sample rate is a compromise: a low sample rate minimizes jitter, while a high sample rate does a better job of capturing detail (e.g. rapid fluctuations) in the warp function. A good rule of thumb is to use at most 1,000 to 10,000 samples for the inverse warp function. For example, if the result will be 1 minute of sound, use a sample rate of 3000 samples / 60 seconds = 50 samples/second. Because Nyquist has no advance information about the warp function, the inverse warp function sample rate must be provided as a parameter. When in doubt, just try something and let your ears be the judge.

\texttt{integrate(\textit{signal})} [SAL]
(integrate \textit{signal}) [LISP]
Computes the integral of \textit{signal}. The start time, sample rate, etc. are taken from \textit{signal}.

\texttt{slope(\textit{signal})} [SAL]
(slope \textit{signal}) [LISP]
Computes the first derivative (slope) of \textit{signal}. The start time, sample rate, etc. are taken from \textit{signal}.

\textbf{7.2.2.1 Oscillators}
**osc**:\(\text{osc}(\text{pitch}[, \text{duration}, \text{table}, \text{phase}])\) [SAL]

\((\text{osc} \text{ pitch} [\text{duration table phase}])\) [LISP]

Returns a sound which is the \text{table} oscillated at \text{pitch} for the given \text{duration}, starting with the \text{phase} (in degrees). Defaults are: \text{duration} 1.0 (second), \text{table} *\text{table}*, \text{phase} 0.0. The default value of *\text{table}* is a sinusoid. Duration is stretched by *\text{warp}* and *\text{sustain}*, amplitude is nominally 1, but scaled by *\text{loudness}*, the start time is logical time 0, transformed by *\text{warp}*, and the sample rate is *\text{sound-srate}*. The effect of time-warping is to warp the starting and ending times only; the signal has a constant unwarped frequency.

\textbf{Note 1:} A \text{table} is 3-element list. See \text{maketable} for a detailed description.

\textbf{Note 2:} in the current implementation, it is assumed that the output should be periodic. See \text{snd-down} and \text{snd-up} for resampling one-shot sounds to a desired sample rate. A future version of \text{osc} will handle both cases.

\textbf{Note 3:} When \text{osc} is called, memory is allocated for the \text{table}, and samples are copied from the sound (the first element of the list which is the \text{table} parameter) to the memory. Every instance of \text{osc} has a private copy of the \text{table}, so the total storage can become large in some cases, for example in granular synthesis with many instances of \text{osc}. In some cases, it may make sense to use \text{snd-flatten} (see Section 7.1.3) to cause the sound to be fully realized, after which the \text{osc} and its \text{table} memory can be reclaimed by garbage collection. The \text{partial} function (see below) does not need a private \text{table} and does not use much space.

**partial**:\(\text{partial}(\text{pitch}, \text{env})\) [SAL]

\((\text{partial} \text{ pitch env})\) [LISP]

Returns a sinusoid at the indicated pitch; the sound is multiplied by \text{env}. The start time and duration are taken from \text{env}, which is of course subject to transformations. The sample rate is *\text{sound-srate}*. The \text{partial} function is faster than \text{osc}.

**sine**:\(\text{sine}(\text{pitch}[, \text{duration}])\) [SAL]

\((\text{sine} \text{ pitch} [\text{duration}])\) [LISP]

Returns a sinusoid at the indicated pitch. The sample rate is *\text{sound-srate}*. This function is like \text{osc} with respect to transformations. The \text{sine} function is faster than \text{osc}.

**hzosc**:\(\text{hzosc}(\text{hz}[, \text{table}, \text{phase}])\) [SAL]

\((\text{hzosc} \text{ hz} [\text{table phase}])\) [LISP]

Returns a sound which is the \text{table} oscillated at \text{hz} starting at \text{phase} degrees. The default \text{table} is *\text{table}* and the default \text{phase} is 0.0 degrees. The default duration is 1.0, but this is stretched as in \text{osc} (see above). The \text{hz} parameter may be a \text{SOUND}, in which case the duration of the result is the duration of \text{hz}. The sample rate is *\text{sound-srate}*. 
osc-saw(hz) [SAL]

(osc-saw hz) [LISP]

Returns a sawtooth waveshape at the indicated frequency (in Hertz). The sample rate is *sound-srate*. The hz parameter may be a sound as in hzosc (see above).

osc-tri(hz) [SAL]

(osc-tri hz) [LISP]

Returns a triangle waveshape at the indicated frequency (in Hertz). The sample rate is *sound-srate*. The hz parameter may be a sound as in hzosc (see above).

osc-pulse(hz, bias [, compare-shape]) [SAL]

(osc-pulse hz bias [compare-shape]) [LISP]

Returns a square pulse with variable width at the indicated frequency (in Hertz). The bias parameter controls the pulse width and should be between -1 and +1, giving a pulse width from 0% (always at -1) to 100% (always at +1). When bias is zero, a square wave is generated. Bias may be a SOUND to create varying pulse width. If bias changes rapidly, strange effects may occur. The optional compare-shape defaults to a hard step at zero, but other shapes may be used to achieve non-square pulses. The osc-pulse behavior is written in terms of other behaviors and defined in the file nyquist.lsp using just a few lines of code. Read the code for the complete story.

amosc(pitch, modulation [, table, phase]) [SAL]

(amosc pitch modulation [table phase]) [LISP]

Returns a sound which is table oscillated at pitch. The output is multiplied by modulation for the duration of the sound modulation. osc-table defaults to *table*, and phase is the starting phase (default 0.0 degrees) within osc-table. The sample rate is *sound-srate*.

fmosc(pitch, modulation [, table, phase]) [SAL]

(fmosc pitch modulation [table phase]) [LISP]

Returns a sound which is table oscillated at pitch plus modulation for the duration of the sound modulation. osc-table defaults to *table*, and phase is the starting phase (default 0.0 degrees) within osc-table. The modulation is expressed in hz, e.g. a sinusoid modulation signal with an amplitude of 1.0 (2.0 peak to peak), will cause a +/- 1.0 hz frequency deviation in sound. Negative frequencies are correctly handled. The sample rate is *sound-srate*.

fmfb(pitch, index [, dur]) [SAL]

(fmfb pitch index [dur]) [LISP]

Returns a sound generated by feedback FM synthesis. The pitch parameter (given in the usual half-step units) controls the fundamental frequency. The index is the amount of feedback, which may be a SOUND or a FLONUM. If index is a FLONUM, dur must be provided (a FLONUM) to specify the duration. Otherwise, dur is ignored if present and the duration is determined by that of index. The sample rate is *sound-srate*. A sinusoid table is used. If index is below 1.1, this generates a sawtooth-like waveform.
buzz\( (n, \textit{pitch}, \textit{modulation}) \) [SAL]  
(buzz \( n \ \textit{pitch} \ \textit{modulation} \) ) [LISP]  
Returns a sound with \( n \) harmonics of equal amplitude and a total amplitude of 1.0, using a well-known function of two cosines. If \( n \) (an integer) is less than 1, it is set to 1. Aliasing will occur if \( n \) is too large. The duration is determined by the duration of the sound \( \textit{modulation} \), which is a frequency modulation term expressed in Hz (see Section 7.2.2.1). Negative frequencies are correctly handled. The sample rate is \(*\text{sound-srate}*\).

pluck\( (\textit{pitch} [, \textit{duration}, \textit{final-amplitude}]) \) [SAL]  
(pluck \( \textit{pitch} \ [\textit{duration} \ \textit{final-amplitude}] \) ) [LISP]  
Returns a sound at the given \( \textit{pitch} \) created using a modified Karplus-Strong plucked string algorithm. The tone decays from an amplitude of about 1.0 to about \( \textit{final-amplitude} \) in \( \textit{duration} \) seconds. The default values are to decay to 0.001 (-60dB) in 1 second. The sample rate is \(*\text{sound-srate}*\).

siosc\( (\textit{pitch}, \textit{modulation}, \textit{tables}) \) [SAL]  
(siosc \( \textit{pitch} \ \textit{modulation} \ \textit{tables} \) ) [LISP]  
Returns a sound constructed by interpolating through a succession of periodic waveforms. The frequency is given (in half steps) by \( \textit{pitch} \) to which a \( \textit{modulation} \) signal (in hz) is added, exactly as in \texttt{fmosc}. The \textit{tables} specify a list of waveforms as follows: \((\textit{table0 time1 table2} \ldots \textit{timeN tableN})\), where each \textit{table} is a sound representing one period. Each \textit{time} is a time interval measured from the starting time. The time is scaled by the nominal duration (computed using \texttt{(local-to-global (get-sustain))}) to get the actual time. Note that this implies linear stretching rather than continuous timewarping of the interpolation or the breakpoints. The waveform is \textit{table0} at the starting time, \textit{table1} after \( \textit{time1} \) (scaled as described), and so on. The duration and logical stop time is given by \textit{modulation}. If \textit{modulation} is shorter than \textit{timeN}, then the full sequence of waveforms is not used. If \textit{modulation} is longer than \textit{timeN}, \textit{tableN} is used after \textit{timeN} without further interpolation.

sampler\( (\textit{pitch}, \textit{modulation} [, \textit{sample}, \textit{npoints}] ) \) [SAL]  
(sampler \( \textit{pitch} \ \textit{modulation} \ [\textit{sample} \ \textit{npoints}] \) ) [LISP]  
Returns a sound constructed by reading a sample from beginning to end and then splicing on copies of the same sound from a loop point to the end. The \textit{pitch} and \textit{modulation} parameters are used as in \texttt{fmosc} described above. The optional \textit{sample} (which defaults to 2048-point sinusoid) is a list of the form

\[(\textit{sound} \ \textit{pitch-number} \ \textit{loop-start})\]

where the first element is a sound containing the sample, the second is the pitch of the sample, and the third element is the time of the loop point. If the loop point is not in the bounds of the sound, it is set to zero. The optional \textit{npoints} specifies how many points should be used for sample interpolation. Currently this parameter defaults to 2 and only 2-point (linear) interpolation is implemented. It is an error to modulate such that the frequency is negative. Note also that the loop point may be fractional. The sample rate is \(*\text{sound-srate}*\).
7.2.2.2 Piece-wise Approximations

There are a number of related behaviors for piece-wise approximations to functions. The simplest of these, pwl was mentioned earlier in the manual. It takes a list of breakpoints, assuming an initial point at (0, 0), and a final value of 0. An analogous piece-wise exponential function, pwe, is provided. Its implicit starting and stopping values are 1 rather than 0. Each of these has variants. You can specify the initial and final values (instead of taking the default). You can specify time in intervals rather than cumulative time. Finally, you can pass a list rather than an argument list. This leads to 16 versions, as shown in Figure 7.2.

All of these functions are implemented in terms of pwl (see nyquist.lsp for the implementations. There are infinite opportunities for errors in these functions: if you leave off a data point, try to specify points in reverse order, try to create an exponential that goes to zero or negative values, or many other bad things, the behavior is not well-defined. Nyquist should not crash, but Nyquist does not necessarily attempt to report errors at this time.

pwl(t₁, l₁, t₂, l₂, ... tₙ) [SAL]
(pwl t₁ l₁ t₂ l₂ ... tₙ) [LISP]
Creates a piece-wise linear envelope with breakpoints at (0, 0), (t₁, l₁), (t₂, l₂), ... (tₙ, 0). The breakpoint times are scaled linearly by the value of *sustain* (if *sustain* is a SOUND, it is evaluated once at the starting time of the envelope). Each breakpoint time is then mapped according to *warp*. The result is a linear interpolation (unwarped) between the breakpoints. The sample rate is *control-srate*.

Breakpoint times are quantized to the nearest sample time. If you specify one or more breakpoints within one sample period, pwl attempts to give a good approximation to the specified function. In particular, if two breakpoints are simultaneous, pwl will move one of them to an adjacent sample, producing a steepest possible step in the signal. The exact details of this "breakpoint munging" is subject to change in future versions. Please report any cases where breakpoint lists give unexpected behaviors. The author will try to apply the "principle of least surprise" to the design. Note that the times are relative to 0; they are not durations of each envelope segment.

pwl-list(breakpoints) [SAL]
(pwl-list breakpoints) [LISP]
If you have a list of breakpoints, you can use apply to apply the pwl function to the breakpoints, but if the list is very long (hundreds or thousands of points), you might get a stack overflow because XLISP has a fixed-size argument stack. Instead, call pwl-list, passing one argument, the list of breakpoints.

pwlv(l₁, t₂, l₂, t₃, l₃, ... tₙ, lₙ) [SAL]
(pwlv l₁ t₂ l₂ t₃ l₃ ... tₙ lₙ) [LISP]
Creates a piece-wise linear envelope with breakpoints at (0, l₁), (t₂, l₂), etc., ending with (tₙ, lₙ). Otherwise, the behavior is like that of pwl.

pwlv-list(breakpoints) [SAL]
(pwlv-list breakpoints) [LISP]
A version of pwlv that takes a single list of breakpoints as its argument. See pwl-list above for the rationale.
Nyquist Reference Manual Chapter 7. Nyquist Functions

Piece-wise Linear Functions:
  Cummulative Time:
    Default initial point at (0, 0), final value at 0:
      pwl
      pwl-list
    Explicit initial value:
      pwlv
      pwlv-list
  Relative Time:
    Default initial point at (0, 0), final value at 0:
      pwlr
      pwlr-list
    Explicit initial value:
      pwlr
      pwlr-list

Piece-wise Exponential Functions:
  Cummulative Time:
    Default initial point at (0, 1), final value at 1:
      pwe
      pwe-list
    Explicit initial value:
      pwev
      pwev-list
  Relative Time:
    Default initial point at (0, 1), final value at 1:
      pwe
      pwe-list
    Explicit initial value:
      pwev
      pwev-list

Figure 7.2: Variants of piece-wise functions.
Nyquist Reference Manual

Chapter 7. Nyquist Functions

pwlr(\(i_1, l_1, i_2, l_2, \ldots i_n\)) [SAL]
(pwlr \(i_1 l_1 i_2 l_2 \ldots i_n\)) [LISP]

Creates a piece-wise linear envelope with breakpoints at \((0, 0), (t_1, l_1), (t_2, l_2), \ldots (t_n, 0)\), where \(t_j\) is the sum of \(i_j\) through \(i_j\). In other words, the breakpoint times are specified in terms of intervals rather than cumulative time. Otherwise, the behavior is like that of \(pwl\).

pwlr-list(breakpoints) [SAL]
(pwlr-list breakpoints) [LISP]

A version of \(pwlr\) that takes a single list of breakpoints as its argument. See \(pwl\)-list above for the rationale.

pwlr\(r(l_1, i_2, l_2, i_3, i_3, \ldots i_n, l_n)\) [SAL]
(pwlr\(r l_1 i_2 l_2 i_3 i_3 \ldots i_n l_n\)) [LISP]

Creates a piece-wise linear envelope with breakpoints at \((0, l_1), (t_2, l_2), \ldots (t_n, l_n)\), where \(t_j\) is the sum of \(i_2\) through \(i_j\). In other words, the breakpoint times are specified in terms of intervals rather than cumulative time. Otherwise, the behavior is like that of \(pwlv\).

pwlr\(r\)-list(breakpoints) [SAL]
(pwlr\(r\)-list breakpoints) [LISP]

A version of \(pwlr\) that takes a single list of breakpoints as its argument. See \(pwl\)-list above for the rationale.

pwe\((t_1, l_1, t_2, l_2, \ldots t_n)\) [SAL]
(pwe \(t_1 l_1 t_2 l_2 \ldots t_n\)) [LISP]

Creates a piece-wise exponential envelope with breakpoints at \((0, 1), (t_1, l_1), (t_2, l_2), \ldots (t_n, 1)\). Exponential segments mean that the ratio of values from sample to sample is constant within the segment. (The current implementation actually takes the log of each value, computes a piece-wise exponential from the points using \(pwl\), then exponentiates each resulting sample. A faster implementation is certainly possible!) Breakpoint values \((l_j)\) must be greater than zero. Otherwise, this function is similar to \(pwl\), including stretch by \(*\text{sustain}*\), mapping according to \(*\text{warp}*\), sample rate based on \(*\text{control-srate}*\), and "breakpoint munging" (see \(pwl\) described above). Default initial and final values are of dubious value with exponentials. See \(pwev\) below for the function you are probably looking for.

pwe\(-list(breakpoints)\) [SAL]
(pwe\(-list breakpoints)\) [LISP]

A version of \(pwe\) that takes a single list of breakpoints as its argument. See \(pwl\)-list above for the rationale.

pwe\(v(l_1, t_2, l_2, t_3, t_3, \ldots t_n, l_n)\) [SAL]
(pwe\(v l_1 t_2 l_2 t_3 t_3 \ldots t_n l_n\)) [LISP]

Creates a piece-wise exponential envelope with breakpoints at \((0, l_1), (t_2, l_2), \ldots (t_n, l_n)\). Otherwise, the behavior is like that of \(pwe\).

pwe\(v\)-list(breakpoints) [SAL]
(pwe\(v\)-list breakpoints) [LISP]

A version of \(pwev\) that takes a single list of breakpoints as its argument. See \(pwl\)-list above for the
**rationale.**

\[ p\text{ower}(i_1, l_1, i_2, l_2, \ldots, i_n) \] [SAL]

(\texttt{p\text{ower} } i_1 \ l_1 \ i_2 \ l_2 \ldots \ i_n) \quad \text{[LISP]}

Creates a piece-wise exponential envelope with breakpoints at \((0, 1), (t_1, l_1), (t_2, l_2), \ldots (t_n, 1)\), where \(t_j\) is the sum of \(i_j\) through \(i_j\). In other words, the breakpoint times are specified in terms of intervals rather than cumulative time. Otherwise, the behavior is like that of \texttt{p\text{we}}. Consider using \texttt{p\text{we}v} instead of this one.

\[ p\text{ower-\text{list}}(\text{breakpoints}) \] [SAL]

(\texttt{p\text{ower-\text{list} } \text{breakpoints}}) \quad \text{[LISP]}

A version of \texttt{p\text{ower}} that takes a single list of breakpoints as its argument. See \texttt{p\text{wl-\text{list}}} above for the rationale.

\[ p\text{e\text{vr}}(l_1, i_2, l_2, i_3, l_3, \ldots, i_n, l_n) \] [SAL]

(\texttt{p\text{e\text{vr} } l_1 \ i_2 \ l_2 \ i_3 \ l_3 \ldots \ i_n \ l_n}) \quad \text{[LISP]}

Creates a piece-wise exponential envelope with breakpoints at \((0, l_1), (t_2, l_2), \ldots, (t_n, l_n)\), where \(t_j\) is the sum of \(i_j\) through \(i_j\). In other words, the breakpoint times are specified in terms of intervals rather than cumulative time. Otherwise, the behavior is like that of \texttt{p\text{we\text{v}}}. Note that this is similar to the \texttt{csound GEN05} generator. Which is uglier, \texttt{GEN05} or \texttt{p\text{e\text{vr}}}?

\[ p\text{e\text{vr}-\text{list}}(\text{breakpoints}) \] [SAL]

(\texttt{p\text{e\text{vr}-\text{list} } \text{breakpoints}}) \quad \text{[LISP]}

A version of \texttt{p\text{e\text{vr}}} that takes a single list of breakpoints as its argument. See \texttt{p\text{wl-\text{list}}} above for the rationale.

### 7.2.2.3 Filter Behaviors

\[ \text{alpass}(\text{\texttt{sound}}, \text{decay}, hz [, minhz]) \] [SAL]

(\texttt{\text{al\text{pass} } \text{\texttt{sound} } \text{decay} \ hz} \ [\text{\texttt{minhz}}]) \quad \text{[LISP]}

Applies an all-pass filter to \texttt{sound}. This all-pass filter creates a delay effect without the resonances of a comb filter. The decay time of the filter is given by \texttt{decay}. The \texttt{hz} parameter must be a number or sound greater than zero. It is used to compute delay, which is then rounded to the nearest integer number of samples (so the frequency is not always exact. Higher sampling rates yield better delay resolution.) The \texttt{decay} may be a sound or a number. In either case, it must also be positive. (Implementation note: an exponentiation is needed to convert \texttt{decay} into the \texttt{feedback} parameter, and exponentiation is typically more time-consuming than the filter operation itself. To get high performance, provide \texttt{decay} at a low sample rate.) The resulting sound will have the start time, sample rate, etc. of \texttt{sound}. If \texttt{hz} is of type \texttt{SOUND}, the delay may be time-varying. Linear interpolation is then used for fractional sample delay, but it should be noted that linear interpolation implies a low-pass transfer function. Thus, this filter may behave differently with a constant \texttt{SOUND} than it does with a \texttt{FLONUM} value for \texttt{hz}. In addition, if \texttt{hz} is of type \texttt{SOUND}, then \texttt{minhz} is required. The \texttt{hz} parameter will be clipped to be greater than \texttt{minhz}, placing an upper bound on the delay buffer length.
**comb**(*sound*, *decay*, *hz*) [SAL]

Applies a comb filter to *sound*. A comb filter emphasizes (resonates at) frequencies that are multiples of a *hz*. The decay time of the resonance is given by *decay*. This is a variation on **feedback-delay** (see below). The *hz* parameter must be a number greater than zero. It is used to compute delay, which is then rounded to the nearest integer number of samples (so the frequency is not always exact. Higher sampling rates yield better delay resolution.) The *decay* may be a sound or a number. In either case, it must also be positive. (Implementation note: an exponentiation is needed to convert *decay* into the *feedback* parameter for **feedback-delay**, and exponentiation is typically more time-consuming than the filter operation itself. To get high performance, provide *decay* at a low sample rate.) The resulting sound will have the start time, sample rate, etc. of *sound*.

**congen**(*gate*, *risetime*, *falltime*) [SAL]

**congen gate risetime falltime** [LISP]

Implements an analog synthesizer-style contour generator. The input *gate* normally goes from 0.0 to 1.0 to create an attack and from 1.0 to 0.0 to start a release. During the attack (output is increasing), the output converges half-way to *gate* in *risetime* (a FLONUM) seconds. During the decay, the half-time is *falltime* seconds. The sample rate, start time, logical stop, and terminate time all come from *gate*. If you want a nice decay, be sure that the *gate* goes to zero and stays there for awhile before *gate* terminates, because congen (and all Nyquist sounds) go immediately to zero at termination time. For example, you can use **pwl** to build a pulse followed by some zero time:

```lisp
(pwl 0 1 duty 1 duty 0 1)
```

Assuming *duty* is less than 1.0, this will be a pulse of duration *duty* followed by zero for a total duration of 1.0.

```lisp
(congen (pwl 0 1 duty 1 duty 0 1) 0.01 0.05)
```

will have a duration of 1.0 because that is the termination time of the pwl input. The decaying release of the resulting envelope will be truncated to zero at time 1.0. (Since the decay is theoretically infinite, there is no way to avoid truncation, although you could multiply by another envelope that smoothly truncates to zero in the last millisecond or two to get both an exponential decay and a smooth final transition to zero.)

**convolve**(*sound*, *response*) [SAL]

**convolve sound response** [LISP]

Convolves two signals. The first can be any length, but the total space required is proportional to the length of *response*. The start time, logical stop time, and sample rate of the output match those of the input *sound*. The physical stop time of the result is the physical stop time of *sound* plus the duration of the *response* so that the result sound includes the "tail" of the filter response. The response is linearly interpolated if necessary to have same sample rate as *sound*. The current implementation uses a "fast convolution" algorithm with a maximum FFT size of 64K, which after zero padding allows up to 32K point convolutions. If the impulse response is longer, it is broken into multiple blocks of 32K samples. There is no limit on the length of the impulse response. This is an Order(N x M) algorithm where N and
M are the number of 32K sample blocks in the sound and response, respectively. Further discussion and examples can be found in demos/convolution.htm.

```
feedback-delay(sound, delay, feedback) [SAL]
(feedback-delay sound delay feedback) [LISP]
```
Applies feedback delay to sound. The delay must be a number (in seconds). It is rounded to the nearest sample to determine the length of the delay. The sample rate is the maximum from sound and feedback (if feedback is also a sound). The amount of feedback should be less than one to avoid an exponential increase in amplitude. The start time and stop time, and logical stop time are taken from sound. Since output is truncated at the stop time of sound, you may want to append some silence to sound to give the filter time to decay.

```
lp(sound, cutoff) [SAL]
(lp sound cutoff) [LISP]
```
Filters sound using a first-order Butterworth low-pass filter. Cutoff may be a float or a signal (for time-varying filtering) and expresses hertz. Filter coefficients (requiring trig functions) are recomputed at the sample rate of cutoff. The resulting sample rate, start time, etc. are taken from sound.

```
tone(sound, cutoff) [SAL]
tone sound cutoff) [LISP]
```
No longer defined; use lp instead, or define it by adding (setfn tone lp) to your program.

```
hp(sound, cutoff) [SAL]
hp sound cutoff) [LISP]
```
Filters sound using a first-order Butterworth high-pass filter. Cutoff may be a float or a signal (for time-varying filtering) and expresses hertz. Filter coefficients (requiring trig functions) are recomputed at the sample rate of cutoff. This filter is an exact complement of lp.

```
atone(sound, cutoff) [SAL]
atone sound cutoff) [LISP]
```
No longer defined; use hp instead, or define it by adding (setfn atone hp) to your program.

```
reson(sound, center, bandwidth [, n]) [SAL]
(reson sound center bandwidth [n]) [LISP]
```
Apply a resonating filter to sound with center frequency center (in hertz), which may be a float or a signal. Bandwidth is the filter bandwidth (in hertz), which may also be a signal. Filter coefficients (requiring trig functions) are recomputed at each new sample of either center or bandwidth, and coefficients are not interpolated. The last parameter n specifies the type of normalization as in Csound: A value of 1 specifies a peak amplitude response of 1.0; all frequencies other than hz are attenuated. A value of 2 specifies the overall RMS value of the amplitude response is 1.0; thus filtered white noise would retain the same power. A value of zero specifies no scaling. The resulting sample rate, start time, etc. are taken from sound.
One application of reson is to simulate resonances in the human vocal tract. See demos/voice_synthesis.htm for sample code and documentation.

areson(sound, center, bandwidth [, n]) [SAL]  
(areson sound center bandwidth [n]) [LISP]  
The areson filter is an exact complement of reson such that if both are applied to the same signal with the same parameters, the sum of the results yeilds the original signal.

shape(signal, table, origin) [SAL]  
(shape signal table origin) [LISP]  
A waveshaping function. Use table as a function; apply the function to each sample of signal to yield a new sound. Signal should range from -1 to +1. Anything beyond these bounds is clipped. Table is also a sound, but it is converted into a lookup table (similar to table-lookup oscillators). The origin is a FLONUM and gives the time which should be considered the origin of table. (This is important because table cannot have values at negative times, but signal will often have negative values. The origin gives an offset so that you can produce suitable tables.) The output at time $t$ is:

$$table(\text{origin} + \text{clip}(signal(t)))$$

where $\text{clip}(x) = \max(1, \min(-1, x))$. (E.g. if table is a signal defined over the interval [0, 2], then origin should be 1.0. The value of table at time 1.0 will be output when the input signal is zero.) The output has the same start time, sample rate, etc. as signal. The shape function will also accept multichannel signals and tables.

Further discussion and examples can be found in demos/distortion.htm. The shape function is also used to map frequency to amplitude to achieve a spectral envelope for Shepard tones in demos/shepard.lsp.

biquad(signal, b0, b1, b2, a0, a1, a2) [SAL]  
(biquad signal b0 b1 b2 a0 a1 a2) [LISP]  
A fixed-parameter biquad filter. All filter coefficients are FLONUMs. See also lowpass2, highpass2, bandpass2, notch2, allpass2, eq-lowshelf, eq-highshelf, eq-band, lowpass4, lowpass6, highpass4, and highpass8 in this section for convenient variations based on the same filter. The equations for the filter are: $z_n = s_n + a_1 * z_{n-1} + a_2 * z_{n-2}$, and $y_n = z_n * b_0 + z_{n-1} * b_1 + z_{n-2} * b_2$.

biquad-m(signal, b0, b1, b2, a0, a1, a2) [SAL]  
(biquad-m signal b0 b1 b2 a0 a1 a2) [LISP]  
A fixed-parameter biquad filter with Matlab sign conventions for a0, a1, and a2. All filter coefficients are FLONUMs.

lowpass2(signal, hz [, q]) [SAL]  
(lowpass2 signal hz [q]) [LISP]  
A fixed-parameter, second-order lowpass filter based on snd-biquad. The cutoff frequency is given by hz (a FLONUM) and an optional Q factor is given by q (a FLONUM).
Nyquist Reference Manual Chapter 7. Nyquist Functions

highpass2(signal, hz [, q]) [SAL]

(highpass2 signal hz [q]) [LISP]

A fixed-parameter, second-order highpass filter based on snd-biquad. The cutoff frequency is given by hz (a FLONUM) and an optional Q factor is given by q (a FLONUM).

bandpass2(signal, hz [, q]) [SAL]

(bandpass2 signal hz [q]) [LISP]

A fixed-parameter, second-order bandpass filter based on snd-biquad. The center frequency is given by hz (a FLONUM) and an optional Q factor is given by q (a FLONUM).

notch2(signal, hz [, q]) [SAL]

(notch2 signal hz [q]) [LISP]

A fixed-parameter, second-order notch filter based on snd-biquad. The center frequency is given by hz (a FLONUM) and an optional Q factor is given by q (a FLONUM).

allpass2(signal, hz [, q]) [SAL]

(allpass2 signal hz [q]) [LISP]

A fixed-parameter, second-order allpass filter based on snd-biquad. The frequency is given by hz (a FLONUM) and an optional Q factor is given by q (a FLONUM).

eq-lowshelf(signal, hz, gain [, slope]) [SAL]

(eq-lowshelf signal hz gain [slope]) [LISP]

A fixed-parameter, second-order bass shelving equalization (EQ) filter based on snd-biquad. The hz parameter (a FLONUM) is the halfway point in the transition, and gain (a FLONUM) is the bass boost (or cut) in dB. The optional slope (a FLONUM) is 1.0 by default, and response becomes peaky at values greater than 1.0.

eq-highshelf(signal, hz, gain [, slope]) [SAL]

(eq-highshelf signal hz gain [slope]) [LISP]

A fixed-parameter, second-order treble shelving equalization (EQ) filter based on snd-biquad. The hz parameter (a FLONUM) is the halfway point in the transition, and gain (a FLONUM) is the treble boost (or cut) in dB. The optional slope (a FLONUM) is 1.0 by default, and response becomes peaky at values greater than 1.0.

eq-band(signal, hz, gain, width) [SAL]

(eq-band signal hz gain width) [LISP]

A fixed- or variable-parameter, second-order midrange equalization (EQ) filter based on snd-biquad, eq-band-ccc and eq-band-vvv. The hz parameter (a FLONUM) is the center frequency, gain (a FLONUM) is the boost (or cut) in dB, and width (a FLONUM) is the half-gain width in octaves. Alternatively, hz, gain, and width may be SOUNDS, but they must all have the same sample rate, e.g. they should all run at the control rate or at the sample rate.
lowpass4(signal, hz) [SAL]
(lowpass4 signal hz) [LISP]
   A four-pole Butterworth lowpass filter. The cutoff frequency is hz (a FLONUM).

lowpass6(signal, hz) [SAL]
(lowpass6 signal hz) [LISP]
   A six-pole Butterworth lowpass filter. The cutoff frequency is hz (a FLONUM).

lowpass8(signal, hz) [SAL]
(lowpass8 signal hz) [LISP]
   An eight-pole Butterworth lowpass filter. The cutoff frequency is hz (a FLONUM).

highpass4(signal, hz) [SAL]
(highpass4 signal hz) [LISP]
   A four-pole Butterworth highpass filter. The cutoff frequency is hz (a FLONUM).

highpass6(signal, hz) [SAL]
(highpass6 signal hz) [LISP]
   A six-pole Butterworth highpass filter. The cutoff frequency is hz (a FLONUM).

highpass8(signal, hz) [SAL]
(highpass8 signal hz) [LISP]
   An eight-pole Butterworth highpass filter. The cutoff frequency is hz (a FLONUM).

tapv(sound, offset, vardelay, maxdelay) [SAL]
(tapv sound offset vardelay maxdelay) [LISP]
   A delay line with a variable position tap. Identical to snd-tapv. See it for details (page 116).

7.2.2.4 Effects

nrev(sound, decay, mix) [SAL]
(nrev sound decay mix) [LISP]

jcrev(sound, decay, mix) [SAL]
(jcrev sound decay mix) [LISP]

prcrev(sound, decay, mix) [SAL]
(prcrev sound decay mix) [LISP]
   These reverbs (nrev, jcrev, and prcrev) are implemented in STK (running within Nyquist). nrev derives from Common Music’s NRev, which consists of 6 comb filters followed by 3 allpass filters, a lowpass filter, and another allpass in series followed by two allpass filters in parallel. jcrev is the
John Chowning reverberator which is based on the use of networks of simple allpass and comb delay filters. This reverb implements three series allpass units, followed by four parallel comb filters, and two decorrelation delay lines in parallel at the output. prcrev is a Perry Cook’s reverberator which is based on the Chowning/Moorer/Schroeder reverberators using networks of simple allpass and comb delay filters. This one implements two series allpass units and two parallel comb filters. The sound input may be single or multi-channel. The decay time is in seconds, and mix sets the mixture of input sound reverb sound, where 0.0 means input only (dry) and 1.0 means reverb only (wet).

stkchorus(sound, depth, freq, mix [, delay]) [SAL]
(stkchorus sound depth freq mix [delay]) [LISP]
Chorus implemented in STK. The input sound can be single or multi-channel. The FLONUM parameters depth and freq set the modulation depth from 0 to 1 and modulation frequency (in Hz), and mix sets the mixture of input sound and chorused sound, where 0.0 means input sound only (dry) and 1.0 means chorused sound only (wet). The parameter delay is a FIXNUM representing the median desired delay length in samples.

pitshift(sound, shift, mix) [SAL]
(pitshift sound shift mix) [LISP]
A pitch shifter implemented in STK. The input sound, a single-channel or multi-channel SOUND is pitch-shifted by shift, a FLONUM ratio. A value of 1.0 means no shift. The parameter mix sets the mixture of input and shifted sounds. A value of 0.0 means input only (dry) and a value of 1.0 means shifted sound only (wet).

7.2.2.5 Physical Models

clarinet(step, breath-env) [SAL]
(clarinet step breath-env) [LISP]
A physical model of a clarinet from STK. The step parameter is a FLONUM that controls the tube length, and the breath-env (a SOUND) controls the air pressure and also determines the length of the resulting sound. The breath-env signal should range from zero to one.

clarinet-freq(step, breath-env, freq-env) [SAL]
(clarinet-freq step breath-env freq-env) [LISP]
A variation of clarinet that includes a variable frequency control, freq-env, which specifies frequency deviation in Hz. The duration of the resulting sound is the minimum duration of breath-env and freq-env. These parameters may be of type FLONUM or SOUND. FLONUMs are coerced into SOUNDs with a nominal duration arbitrarily set to 30.
clarinet-all(step, breath-env, freq-env, vibrato-freq, vibrato-gain, 
reed-stiffness, noise) [SAL]

(clarinet-all step breath-env freq-env vibrato-freq vibrato-gain 
reed-stiffness noise) [LISP]

A variation of clarinet-freq that includes controls vibrato-freq (a FLONUM for vibrato frequency in Hertz), vibrato-gain (a FLONUM for the amount of amplitude vibrato), reed-stiffness (a FLONUM or SOUND controlling reed stiffness in the clarinet model), and noise (a FLONUM or SOUND controlling noise amplitude in the input air pressure). The vibrato-gain is a number from zero to one, where zero indicates no vibrato, and one indicates a plus/minus 50% change in breath envelope values. Similarly, the noise parameter ranges from zero to one where zero means no noise and one means white noise with a peak amplitude of plus/minus 40% of the breath-env. The reed-stiffness parameter varies from zero to one. The duration of the resulting sound is the minimum duration of breath-env, freq-env, reed-stiffness, and noise. As with clarinet-freq, these parameters may be either FLONUMs or SOUNDS, and FLONUMs are coerced to sounds with a nominal duration of 30.

sax(step, breath-env) [SAL]

(sax step breath-env) [LISP]

A physical model of a sax from STK. The step parameter is a FLONUM that controls the tube length, and the breath-env controls the air pressure and also determines the length of the resulting sound. The breath-env signal should range from zero to one.

sax-freq(step, breath-env, freq-env) [SAL]

(sax-freq step breath-env freq-env) [LISP]

A variation of sax that includes a variable frequency control, freq-env, which specifies frequency deviation in Hz. The duration of the resulting sound is the minimum duration of breath-env and freq-env. These parameters may be of type FLONUM or SOUND. FLONUMs are coerced into SOUNDS with a nominal duration arbitrarily set to 30.

sax-all(step, breath-env, freq-env, vibrato-freq, vibrato-gain, 
reed-stiffness, noise, blow-pos, reed-table-offset) [SAL]

(sax-all step breath-env freq-env vibrato-freq vibrato-gain 
reed-stiffness noise blow-pos reed-table-offset) [LISP]

A variation of sax-freq that includes controls vibrato-freq (a FLONUM for vibrato frequency in Hertz), vibrato-gain (a FLONUM for the amount of amplitude vibrato), reed-stiffness (a SOUND controlling reed stiffness in the sax model), noise (a SOUND controlling noise amplitude in the input air pressure), blow-pos (a SOUND controlling the point of excitation of the air column), and reed-table-offset (a SOUND controlling a parameter of the reed model). The vibrato-gain is a number from zero to one, where zero indicates no vibrato, and one indicates a plus/minus 50% change in breath envelope values. Similarly, the noise parameter ranges from zero to one where zero means no noise and one means white noise with a peak amplitude of plus/minus 40% of the breath-env. The reed-stiffness, blow-pos, and reed-table-offset parameters all vary from zero to one. The duration of the resulting sound is the minimum duration of breath-env, freq-env, reed-stiffness, noise, breath-env, blow-pos, and reed-table-offset. As with sax-freq, these parameters may be either FLONUMs or SOUNDS, and FLONUMs are coerced to sounds with a nominal duration of 30.
flute\((\text{step}, \text{breath-env})\) [SAL]

(flute \text{step} \text{breath-env}) [LISP]

A physical model of a flute from STK. The \text{step} parameter is a FLONUM that controls the tube length, and the \text{breath-env} controls the air pressure and also determines the starting time and length of the resulting sound. The \text{breath-env} signal should range from zero to one.

flute-freq\((\text{step}, \text{breath-env}, \text{freq-env})\) [SAL]

(flute-freq \text{step} \text{breath-env} \text{freq-env}) [LISP]

A variation of flute that includes a variable frequency control, \text{freq-env}, which specifies frequency deviation in Hz. The duration of the resulting sound is the minimum duration of \text{breath-env} and \text{freq-env}. These parameters may be of type FLONUM or SOUND. FLONUMs are coerced into SOUNDs with a nominal duration arbitrarily set to 30.

flute-all\((\text{step}, \text{breath-env}, \text{freq-env}, \text{vibrato-freq}, \text{vibrato-gain}, \text{jet-delay}, \text{noise})\) [SAL]

(flute-all \text{step} \text{breath-env} \text{freq-env} \text{vibrato-freq} \text{vibrato-gain} \text{jet-delay} \text{noise}) [LISP]

A variation of clarinet-freq that includes controls \text{vibrato-freq} (a FLONUM for vibrato frequency in Hz), \text{vibrato-gain} (a FLONUM for the amount of amplitude vibrato), \text{jet-delay} (a FLONUM or SOUND controlling jet delay in the flute model), and noise (a FLONUM or SOUND controlling noise amplitude in the input air pressure). The \text{vibrato-gain} is a number from zero to one where zero means no vibrato, and one indicates a plus/minus 50% change in breath envelope values. Similarly, the \text{noise} parameter ranges from zero to one, where zero means no noise and one means white noise with a peak amplitude of plus/minus 40% of the \text{breath-env}. The \text{jet-delay} is a ratio that controls a delay length from the flute model, and therefore it changes the pitch of the resulting sound. A value of 0.5 will maintain the pitch indicated by the \text{step} parameter. The duration of the resulting sound is the minimum duration of \text{breath-env}, \text{freq-env}, \text{jet-delay}, and \text{noise}. These parameters may be either FLONUMs or SOUNDs, and FLONUMs are coerced to sounds with a nominal duration of 30.

bowed\((\text{step}, \text{bowpress-env})\) [SAL]

(bowed \text{step} \text{bowpress-env}) [LISP]

A physical model of a bowed string instrument from STK. The \text{step} parameter is a FLONUM that controls the string length, and the \text{bowpress-env} controls the bow pressure and also determines the duration of the resulting sound. The \text{bowpress-env} signal should range from zero to one.

bowed-freq\((\text{step}, \text{bowpress-env}, \text{freq-env})\) [SAL]

(bowed-freq \text{step} \text{bowpress-env} \text{freq-env}) [LISP]

A variation of bowed that includes a variable frequency control, \text{freq-env}, which specifies frequency deviation in Hz. The duration of the resulting sound is the minimum duration of \text{bowpress-env} and \text{freq-env}. These parameters may be of type FLONUM or SOUND. FLONUMs are coerced into SOUNDs with a nominal duration arbitrarily set to 30s.
mandolin(step, dur, koptional detune) [SAL]
(mandolin step dur detune) [LISP]
   A physical model of a plucked double-string instrument from STK. The step parameter is a FLONUM
   which specifies the desired pitch, dur means the duration of the resulting sound and detune is a FLONUM
   that controls the relative detune of the two strings. A value of 1.0 means unison. The default value is
   4.0. Note: body-size (see snd-mandolin does not seem to work correctly, so a default value is always
   used by mandolin.

wg-uniform-bar(step, bowpress-env) [SAL]
(wg-uniform-bar step bowpress-env) [LISP]

wg-tuned-bar(step, bowpress-env) [SAL]
(wg-tuned-bar step bowpress-env) [LISP]

wg-glass-harm(step, bowpress-env) [SAL]
(wg-glass-harm step bowpress-env) [LISP]

wg-tibetan-bowl(step, bowpress-env) [SAL]
(wg-tibetan-bowl step bowpress-env) [LISP]
   These sounds are presets for a Banded Wave Guide Percussion instrument implemented in STK. The
   parameter step is a FLONUM that controls the resultant pitch, and bowpress-env is a SOUND ranging from
   zero to one that controls a parameter of the model. In addition, bowpress-env determines the duration
   of the resulting sound. (Note: The bowpress-env does not seems influence the timbral quality of the
   resulting sound).

modalbar(preset, step, dur) [SAL]
(modalbar preset step dur) [LISP]
   A physical model of a struck bar instrument implemented in STK. The parameter preset is one of
   the symbols MARIMBA, VIBRAPHONE, AGOGO, WOOD1, RESO, WOOD2, BEATS, TWO-FIXED, or CLUMP. The
   symbol must be quoted, e.g. for SAL syntax use quote(marimba), and for Lisp syntax use 'marimba.
   The parameter step is a FLONUM that sets the pitch (in steps), and dur is the duration in seconds.

sitar(step, dur) [SAL]
(sitar step dur) [LISP]
   A sitar physical model implemented in STK. The parameter step is a FLONUM that sets the pitch, and
   dur is the duration.

7.2.2.6 Phase Vocoder

phasevocoder(s, map, [fftsize, hopsize, mode]) [SAL]
(phasevocoder s map [fftsize hopsize mode]) [LISP]
   Phase vocoder: the input SOUND s is stretched in time according to the control signal (a SOUND)
The `map` parameter must be strictly non-decreasing. It specifies the input time for each time in the output. E.g. if `map` at time 3 is 4, then the output at time 3 will sound like the input at time 4, assuming `s` and `map` start at time 0. In the general case, `s` and `map` must start at the same time. The phase vocoder operation is applied as if `s` and `map` started at time 0, and the result is shifted to start at the same starting time as `s` and `map`. The `fftsize` is a FIXNUM and power of 2, indicating the size of the analysis and synthesis windows in samples. The default is 2048. The `hopsize` is the hop size (in samples, a FIXNUM) of the synthesis stage where overlapping windows are added. The `hopsize` should be a power of 2 that is smaller than the `fftsize`. The default is the `fftsize`/8. Note that the analysis hopsize is determined by the `map`. If the playback speed (slope of `map`) is too great, the analysis hopsize will be large and the sound quality will suffer. You can always make the analysis hopsize smaller by making the synthesis hopsize smaller (at the cost of greater computation). The `mode` is a FIXNUM that defaults to 0, meaning a standard phase vocoder. A value of 1 invokes a phase computation that attempts to reduce phase artifacts by preserving the phase relationships between peaks and nearby bins. A value of 2 invokes a "robot voice" mode that assigns fixed phases and creates a constant pitch (controlled by the hopsize) vocoder-like effect. (Thanks to M.C. Sharma for this idea.)

```lisp
pv-time-pitch(s, stretchfn, pitchfn, dur, [fftsize, hopsize, mode]) [LISP]
```

Combines phase vocoder and resampling to perform simultaneous time stretching and pitch shifting. The input SOUND `s` is stretched according to SOUND `stretchfn` and pitch-shifted according to SOUND `pitchfn`. The approximate output duration should be specified by `dur` (which is used to optimize the sample rate of computed control functions), and the remaining parameters `fftsize`, `hopsize` and `mode` are passed to the phase vocoder (see `phasevocoder` above). The `stretchfn` gives the factor by which the input should be stretched at each point in time; thus, the total duration is the integral of this function. The `pitchfn` specifies the amount by which pitch should be shifted at each point in time. For example, where `pitchfn` is 2, the sample rate will be doubled, increasing pitch and frequencies by an octave. The phase vocoder is used to compensate for time stretching caused by resampling, so `stretchfn` and `pitchfn` operate independently.

### 7.2.2.7 More Behaviors

```lisp
clip(sound, peak) [LISP]
```

Hard limit `sound` to the given `peak`, a positive number. The samples of `sound` are constrained between an upper value of `peak` and a lower value of `-peak`. If `sound` is a number, `clip` will return `sound` limited by `peak`. If `sound` is a multichannel sound, `clip` returns a multichannel sound where each channel is clipped. The result has the type, sample rate, starting time, etc. of `sound`.

```lisp
s-abs(sound) [LISP]
```

A generalized absolute value function. If `sound` is a SOUND, compute the absolute value of each sample. If `sound` is a number, just compute the absolute value. If `sound` is a multichannel sound, return a
multichannel sound with \texttt{s-abs} applied to each element. The result has the type, sample rate, starting time, etc. of \texttt{sound}.

\texttt{s-sqrt(sound)} [SAL]
\texttt{(s-sqrt sound)} [LISP]
A generalized square root function. If \textit{sound} is a \texttt{SOUND}, compute the square root of each sample. If \textit{sound} is a number, just compute the square root. If \textit{sound} is a multichannel sound, return a multichannel sound with \texttt{s-sqrt} applied to each element. The result has the type, sample rate, starting time, etc. of \textit{sound}. In taking square roots, if an input sample is less than zero, the corresponding output sample is zero. This is done because the square root of a negative number is undefined.

\texttt{s-exp(sound)} [SAL]
\texttt{(s-exp sound)} [LISP]
A generalized exponential function. If \textit{sound} is a \texttt{SOUND}, compute $e^x$ for each sample $x$. If \textit{sound} is a number $x$, just compute $e^x$. If \textit{sound} is a multichannel sound, return a multichannel sound with \texttt{s-exp} applied to each element. The result has the type, sample rate, starting time, etc. of \textit{sound}.

\texttt{s-log(sound)} [SAL]
\texttt{(s-log sound)} [LISP]
A generalized natural log function. If \textit{sound} is a \texttt{SOUND}, compute $\ln(x)$ for each sample $x$. If \textit{sound} is a number $x$, just compute $\ln(x)$. If \textit{sound} is a multichannel sound, return a multichannel sound with \texttt{s-log} applied to each element. The result has the type, sample rate, starting time, etc. of \textit{sound}. Note that the \texttt{ln} of 0 is undefined (some implementations return negative infinity), so use this function with care.

\texttt{s-max(sound1, sound2)} [SAL]
\texttt{(s-max sound1 sound2)} [LISP]
Compute the maximum of two functions, \textit{sound1} and \textit{sound2}. This function also accepts numbers and multichannel sounds and returns the corresponding data type. The start time of the result is the maximum of the start times of \textit{sound1} and \textit{sound2}. The logical stop time and physical stop time of the result is the minimum of the logical stop and physical stop times respectively of \textit{sound1} and \textit{sound2}. Note, therefore, that the result value is zero except within the bounds of both input sounds.

\texttt{s-min(sound1, sound2)} [SAL]
\texttt{(s-min sound1 sound2)} [LISP]
Compute the minimum of two functions, \textit{sound1} and \textit{sound2}. This function also accepts numbers and multichannel sounds and returns the corresponding data type. The start time of the result is the maximum of the start times of \textit{sound1} and \textit{sound2}. The logical stop time and physical stop time of the result is the minimum of the logical stop and physical stop times respectively of \textit{sound1} and \textit{sound2}. Note, therefore, that the result value is zero except within the bounds of both input sounds.

\texttt{osc-note(pitch [, duration, env, loud, table])} [SAL]
\texttt{(osc-note pitch [duration env loud table])} [LISP]
Same as \texttt{osc}, but \texttt{osc-note} multiplies the result by \textit{env}. The \textit{env} may be a sound, or a list supplying $(t_1 t_2 t_4 l_1 l_2 l_3)$. The result has a sample rate of *\texttt{sound-srate}*.
quantize(sound, steps) [SAL]
(quantize sound steps) [LISP]
Quantizes sound as follows: sound is multiplied by steps and rounded to the nearest integer. The result is then divided by steps. For example, if steps is 127, then a signal that ranges from -1 to +1 will be quantized to 255 levels (127 less than zero, 127 greater than zero, and zero itself). This would match the quantization Nyquist performs when writing a signal to an 8-bit audio file. The sound may be multi-channel.

ramp([duration]) [SAL]
(ramp [duration]) [LISP]
Returns a linear ramp from 0 to 1 over duration (default is 1). The function actually reaches 1 at duration, and therefore has one extra sample, making the total duration be duration + 1/*Control-srate*. See Figure 7.3 for more detail. Ramp is unaffected by the sustain transformation. The effect of time warping is to warp the starting and ending times only. The ramp itself is unwarped (linear). The sample rate is *control-srate*.

rms(sound [, rate, window-size]) [SAL]
(rms sound [rate window-size]) [LISP]
Computes the RMS of sound using a square window of size window-size. The result has a sample rate of rate. The default value of rate is 100 Hz, and the default window size is 1/rate seconds (converted to samples). The rate is a FLONUM and window-size is a FIXNUM.

recip(sound) [SAL]
(recip sound) [LISP]
A generalized reciprocal function. If sound is a SOUND, compute 1/x for each sample x. If sound is a number x, just compute 1/x. If sound is a multichannel sound, return a multichannel sound with recip applied to each element. The result has the type, sample rate, starting time, etc. of sound. Note that the reciprocal of 0 is undefined (some implementations return infinity), so use this function with care on sounds. Division of sounds is accomplished by multiplying by the reciprocal. Again, be careful not to divide by zero.

s-rest([duration]) [SAL]
(s-rest [duration]) [LISP]
Create silence (zero samples) for the given duration at the sample rate *sound-srate*. Default duration is 1.0 sec, and the sound is transformed in time according to *warp*. Note: rest is a Lisp function that is equivalent to cdr. Be careful to use s-rest when you need a sound!

noise([duration]) [SAL]
(noise duration) [LISP]
Generate noise with the given duration. Duration (default is 1.0) is transformed according to *warp*. The sample rate is *sound-srate* and the amplitude is +/- *loud*.
Figure 7.3: Ramps generated by \texttt{pwl} and \texttt{ramp} functions. The \texttt{pwl} version ramps toward the breakpoint (1, 1), but in order to ramp back to zero at breakpoint (1, 0), the function never reaches an amplitude of 1. If used at the beginning of a \texttt{seq} construct, the next sound will begin at time 1. The \texttt{ramp} version actually reaches breakpoint (1, 1); notice that it is one sample longer than the \texttt{pwl} version. If used in a sequence, the next sound after \texttt{ramp} would start at time $1 + P$, where $P$ is the sample period.

\begin{verbatim}
yin(sound, minstep, maxstep, stepsize) [SAL]
(yin sound minstep maxstep stepsize) [LISP]
\end{verbatim}

Fundamental frequency estimation (pitch detection). Use the YIN algorithm to estimate the fundamental frequency of \texttt{sound}, which must be a \texttt{SOUND}. The \texttt{minstep}, a \texttt{FLONUM}, is the minimum frequency considered (in steps), \texttt{maxstep}, a \texttt{FLONUM}, is the maximum frequency considered (in steps), and \texttt{stepsize}, a \texttt{FIXNUM}, is the desired hop size. The result is a "stereo" signal, i.e. an array of two \texttt{SOUNDS}, both at the same sample rate, which is approximately the sample rate of \texttt{sound} divided by \texttt{stepsize}. The first \texttt{SOUND} consists of frequency estimates (in units of steps, i.e. middle C = 60). The second sound consists of values that measure the confidence or reliability of the frequency estimate. A small value (less than 0.1) indicates fairly high confidence. A larger value indicates lower confidence. This number can also be thought of as a ratio of non-periodic power to periodic power. When the number is low, it means the signal is highly periodic at that point in time, so the period estimate will be reliable. Hint #1: See Alain de Cheveigne and Hideki Kawahara’s article "YIN, a Fundamental Frequency Estimator for Speech and Music" in the Journal of the Acoustic Society of America, April 2002 for details on the yin algorithm. Hint #2: Typically, the \texttt{stepsize} should be at least the expected number of samples in one period so that the fundamental frequency estimates are calculated at a rate far below the sample rate of the signal. Frequency does not change rapidly and the yin algorithm is fairly slow. To optimize speed, you may want to use less than 44.1 kHz sample rates for input sounds. Yin uses interpolation to achieve potentially fractional-sample-accurate estimates, so higher sample rates do not necessarily help the algorithm and definitely slow it down. The computation time is $O(n^2)$ per estimate, where $n$ is the number of samples in the longest period considered. Therefore, each increase of \texttt{minstep} by 12 (an
octave) gives you a factor of 4 speedup, and each decrease of the sample rate of sound by a factor of two gives you another factor of 4 speedup. Finally, the number of estimates is inversely proportional to stepsize. Hint #3: Use snd-srate (see Section 7.1.3) to get the exact sample rate of the result, which will be the sample rate of sound divided by stepsize. E.g. (snd-srate (aref yin-output 0)), where yin-output is a result returned by yin, will be the sample rate of the estimates.

### 7.3 Transformations

These functions change the environment that is seen by other high-level functions. Note that these changes are usually relative to the current environment. There are also "absolute" versions of each transformation function, with the exception of seq, seqrep, sim, and simrep. The "absolute" versions (starting or ending with "abs") do not look at the current environment, but rather set an environment variable to a specific value. In this way, sections of code can be insulated from external transformations.

#### abs-env (beh) [SAL]

(abs-env beh) [LISP]

Compute beh in the default environment. This is useful for computing waveform tables and signals that are "outside" of time. For example, (at 10.0 (abs-env (my-beh))) is equivalent to (abs-env (my-beh)) because abs-env forces the default environment. Or in SAL, we would say abs-env(my-beh()) @ 10 is equivalent to abs-env(my-beh()).

#### at(time, beh) [SAL]

(at time beh) [LISP]

Evaluate beh with *warp* shifted by time. In SAL, you can use the infix operator @ as in beh @ time. To discover how the environment is shifting time, use local-to-global(time). Most commonly, you call local-to-global(0) to find when a sound created in the current environment will start, expressed in absolute (global) terms. This can be regarded as the "current time."

#### at-abs(time, beh) [SAL]

(at-abs time beh) [LISP]

Evaluate beh with *warp* shifted so that local time 0 maps to time. In SAL, you can use the infix operator @@ as in beh @@ time.

#### continuous-control-warp(beh) [SAL]

(continuous-control-warp beh) [LISP]

Applies the current warp environment to the signal returned by beh. The result has the default control sample rate *control-srate*. Linear interpolation is currently used. Implementation: beh is first evaluated without any shifting, stretching, or warping. The result is functionally composed with the inverse of the environment’s warp function.

#### continuous-sound-warp(beh) [SAL]

(continuous-sound-warp beh) [LISP]

Applies the current warp environment to the signal returned by beh. The result has the default sound
sample rate *sound-srate*. Linear interpolation is currently used. See continuous-control-warp for implementation notes.

control-srate-abs(srate, beh) [SAL]
(control-srate-abs srate beh) [LISP]
Evaluate beh with *control-srate* set to sample rate srate. Note: there is no "relative" version of this function.

extract(start, stop, beh) [SAL]
(extract start stop beh) [LISP]
Returns a sound which is the portion of beh between start and stop. Note that this is done relative to the current *warp*. The result is shifted to start according to *warp*, so normally the result will start without a delay of start.

extract-abs(start, stop, beh) [SAL]
(extract-abs start stop beh) [LISP]
Returns a sound which is the portion of beh between start and stop, independent of the current *warp*. The result is shifted to start according to *warp*.

loud(volume, beh) [SAL]
(loud volume beh) [LISP]
Evaluates beh with *loud* incremented by volume. (Recall that *loud* is in decibels, so increment is the proper operation.)

loud-abs(volume, beh) [SAL]
(loud-abs volume beh) [LISP]
Evaluates beh with *loud* set to volume.

sound-srate-abs(srate, beh) [SAL]
(sound-srate-abs srate beh) [LISP]
Evaluate beh with *sound-srate* set to sample rate srate. Note: there is no "relative" version of this function.

stretch(factor, beh) [SAL]
(stretch factor beh) [LISP]
Evaluates beh with *warp* scaled by factor. The effect is to "stretch" the result of beh (under the current environment) by factor. See Chapter 4 for more information. Use get-duration(dur) to get the nominal actual duration of a behavior that locally has a duration of dur. Here, "nominal" means what would be expected if the behavior obeys the shift, stretch, and warp components of the environment. (Any behavior is free to deviate from the nominal timing. For example, a percussion sound might have a fixed duration independent of the stretch factor.) Also, "actual" means global or absolute time, and "locally" means within the environment where get-duration is called. get-duration works by mapping the current time (local time 0) using local-to-global to obtain an actual start time, and mapping dur to obtain an actual end time. The difference is returned.
stretch-abs(factor, beh) [SAL]
(stretch-\texttt{abs factor beh}) [LISP]
   Evaluates beh with \texttt{*warp*} set to a linear time transformation where each unit of logical time maps to factor units of real time. The effect is to stretch the nominal behavior of beh (under the default global environment) by factor. See Chapter 4 for more information.

sustain(factor, beh) [SAL]
(sustain factor beh) [LISP]
   Evaluates beh with \texttt{*sustain*} scaled by factor. The effect is to "stretch" the result of beh (under the current environment) by factor; however, the logical stop times are not stretched. Therefore, the overall duration of a sequence is not changed, and sounds will tend to overlap if \texttt{*sustain*} is greater than one (legato) and be separated by silence if \texttt{*sustain*} is less than one.

sustain-abs(factor, beh) [SAL]
(sustain-abs factor beh) [LISP]
   Evaluates beh with \texttt{*sustain*} set to factor. (See sustain, above.)

transpose(amount, beh) [SAL]
(transpose amount beh) [LISP]
   Evaluates beh with \texttt{*transpose*} shifted by amount. The effect is relative transposition by amount semitones.

transpose-abs(amount, beh) [SAL]
(transpose-abs amount beh) [LISP]
   Evaluates beh with \texttt{*transpose*} set to amount. The effect is the transposition of the nominal pitches in beh (under the default global environment) by amount.

warp(fn, beh) [SAL]
(warp fn beh) [LISP]
   Evaluates beh with \texttt{*warp*} modified by fn. The idea is that beh and fn are written in the same time system, and fn warps that time system to local time. The current environment already contains a mapping from local time to global (real) time. The value of \texttt{*warp*} in effect when beh is evaluated is the functional composition of the initial \texttt{*warp*} with fn.

warp-abs(fn, beh) [SAL]
(warp-abs fn beh) [LISP]
   Evaluates beh with \texttt{*warp*} set to fn. In other words, the current \texttt{*warp*} is ignored and not composed with fn to form the new \texttt{*warp*}.

7.4 Combination and Time Structure

These behaviors combine component behaviors into structures, including sequences (melodies), simultaneous sounds (chords), and structures based on iteration. See also the \texttt{trigger} function, described in Section 9.1.4, which uses a SOUND to trigger instances of a behavior.
seq(beh1 [, beh2, ...]) [SAL]
(seq beh1 [beh2 ...]) [LISP]
    Evaluates the first behavior beh1 according to *time* and each successive behavior at the logical-stop time of the previous one. The results are summed to form a sound whose logical-stop is the logical-stop of the last behavior in the sequence. Each behavior can result in a multichannel sound, in which case, the logical stop time is considered to be the maximum logical stop time of any channel. The number of channels in the result is the number of channels of the first behavior. If other behaviors return fewer channels, new channels are created containing constant zero signals until the required number of channels is obtained. If other behaviors return a simple sound rather than multichannel sounds, the sound is automatically assigned to the first channel of a multichannel sound that is then filled out with zero signals. If another behavior returns more channels than the first behavior, the error is reported and the computation is stopped. Sample rates are converted up or down to match the sample rate of the first sound in a sequence.

seqrep(var, limit, beh) [SAL]
(seqrep var limit beh) [LISP]
    Iteratively evaluates beh with the atom var set with values from 0 to limit-1, inclusive. These sounds are placed sequentially in time as if by seq. The symbol var is a read-only local variable to beh. Assignments are not restricted or detected, but may cause a run-time error or crash. In LISP, the syntax is (seqrep (var limit) beh).

sim([beh1, beh2, ...]) [SAL]
(sim [beh1 beh2 ...]) [LISP]
    Returns a sound which is the sum of the given behaviors evaluated with the current value of *warp*. If behaviors return multiple channel sounds, the corresponding channels are added. If the number of channels does not match, the result has as many channels as the argument with the most channels. For example, if a two-channel sound [L, R] is added to a four-channel sound [C1, C2, C3, C4], the result is [L + C1, R + C2, C3, C4]. Arguments to sim may also be numbers. If all arguments are numbers, sim is equivalent (although slower than) the LISP + function. If a number is added to a sound, snd-offset is used to add the number to each sample of the sound. The result of adding a number to two or more sounds with different durations is not defined. Use const to coerce a number to a sound of a specified duration. An important limitation of sim is that it cannot handle hundreds of behaviors due to a stack size limitation in XLISP. To compute hundreds of sounds (e.g. notes) at specified times, see timed-seq below. See also sum below. Notice that sim is not transitive due to coercion rules: Using SAL syntax, sim(a, sim(b, c)) may not produce the same result as sim(sim(a, b), c).

simrep(var, limit, beh) [SAL]
(simrep var limit beh) [LISP]
    Iteratively evaluates beh with the atom var set with values from 0 to limit-1, inclusive. These sounds are then placed simultaneously in time as if by sim. In LISP, the syntax is (seqrep (var limit) beh).
set-logical-stop(beh, time) [SAL]
(set-logical-stop beh time) [LISP]

Returns a sound with time as the logical stop time.

sum(a [, b, ...]) [SAL]
(sum a [b ...]) [LISP]

Returns the sum of a, b, ..., allowing mixed addition of sounds, multichannel sounds and numbers. Identical to sim. In SAL, use the infix "+" operator. See sim just above for more detail.

mult(a [, b, ...]) [SAL]
(mult a [b ...]) [LISP]

Returns the product of a, b, ..., allowing mixed multiplication of sounds, multichannel sounds and numbers.

diff(a, b) [SAL]
(diff a b) [LISP]

Returns the difference between a and b. This function is defined as (sum a (prod -1 b)).

timed-seq(score) [SAL]
(timed-seq score) [LISP]

Computes sounds from a note list or "score." The score is of the form: ‘((time1 stretch1 beh1) (time2 stretch2 beh2) ...), where timeN is the starting time, stretchN is the stretch factor, and behN is the behavior. Note that score is normally a quoted list! The times must be in increasing order, and each behN is evaluated using Lisp’s eval, so the behN behaviors cannot refer to local parameters or local variables. The advantage of this form over seq is that the behaviors are evaluated one-at-a-time which can take much less stack space and overall memory. One special "behavior" expression is interpreted directly by timed-seq: (SCORE-BEGIN-END) is ignored, not evaluated as a function. Normally, this special behavior is placed at time 0 and has two parameters: the score start time and the score end time. These are used in Xmusic functions. If the behavior has a :pitch keyword parameter which is a list, the list represents a chord, and the expression is replaced by a set of behaviors, one for each note in the chord. It follows that if :pitch is nil, the behavior represents a rest and is ignored.

7.5 Sound File Input and Output

play sound [SAL]
(play sound) [LISP]

Play the sound through the DAC. Note that play is a command in SAL. In XLISP, it is a function, so the syntax is (play sound), and in SAL, you can call the XLISP function as #play(sound). The play command or function writes a file and plays it. The sound is any expression that evaluates to a SOUND. Typically, this should be function call, in which case the samples are computed incrementally and not retained in main memory (an advantage for large sounds). If the expression is a variable
containing a SOUND (which may or may not be fully evaluated yet), the SOUND is fully evaluated and all samples are retained in the variable. The play function is defined in the file system.lsp. The variable *default-sf-dir* names a directory into which to save a sound file. Be careful not to call play or sound-play within a function and then invoke that function from another play command.

By default, Nyquist will try to normalize sounds using the method named by *autonorm-type*, which is 'lookahead by default. The lookahead method precomputes and buffers *autonorm-max-samples* samples, finds the peak value, and normalizes accordingly. The 'previous method bases the normalization of the current sound on the peak value of the (entire) previous sound. This might be good if you are working with long sounds that start rather softly. See Section 5.3 for more details.

If you want precise control over output levels, you should turn this feature off by typing (using SAL syntax):

```
autonorm-off()
```

Reenable the automatic normalization feature by typing:

```
autonorm-on()
```

Play normally produces real-time output. The default is to send audio data to the DAC as it is computed in addition to saving samples in a file. If computation is slower than real-time, output will be choppy, but since the samples end up in a file, you can type (r) to replay the stored sound. Real-time playback can be disabled by (using SAL syntax):

```
sound-off()
```

and reenabled by:

```
sound-on()
```

Disabling real-time playback has no effect on (play-file) or (r).

While sounds are playing, typing control-A to Nyquist (or clicking the Mark button in the NyquistIDE) will push the estimated elapsed audio time onto the head of the list stored in *audio-markers*. Because samples are computed in blocks and because there is latency between sample computation and sample playback, the elapsed time may not be too accurate, and the computed elapsed time may not advance after all samples have been computed but the sound is still playing.

```
play-file(filename) [SAL]
(play-file filename) [LISP]
```

Play the contents of a sound file named by filename. The s-read function is used to read the file, and unless filename specifies an absolute path or starts with ".", it will be read from *default-sf-dir*.

```
autonorm-on() [SAL]
(autonorm-on) [LISP]
```

Enable automatic adjustment of a scale factor applied to sounds computed using the play command.

```
autonorm-off() [SAL]
(autonorm-off) [LISP]
```

Disable automatic adjustment of a scale factor applied to sounds computed using the play command.
sound-on() [SAL]
(sound-on) [LISP]
    Enable real-time audio output when sound is computed by the play command.

sound-off() [SAL]
(sound-off) [LISP]
    Disable real-time audio output when sound is computed by the play command.

s-save(expression, maxlen, filename, format: format, mode: mode, bits: bits,
    swap: flag, play: play) [SAL]
(s-save expression maxlen filename :format format :mode mode :bits bits
    :swap flag :play play) [LISP]
    Evaluates the expression, which should result in a sound or an array of sounds, and writes the result to the given filename. A FLONUM is returned giving the maximum absolute value of all samples written. (This is useful for normalizing sounds and detecting sample overflow.) If play is not NIL, the sound will be output through the computer’s audio output system. (play: [SAL] or :play [LISP] is not implemented on all systems; if it is implemented, and filename is NIL, then this will play the file without also writing a file.) The latency (length of audio buffering) used to play the sound is 0.3s by default, but see snd-set-latency. If a multichannel sound (array) is written, the channels are up-sampled to the highest rate in any channel so that all channels have the same sample rate. The maximum number of samples written per channel is given by maxlen, which allows writing the initial part of a very long or infinite sound. A header is written according to format, samples are encoded according to mode, using bits bits/sample, and bytes are swapped if flag is not NIL. Defaults for these are *default-sf-format*, *default-sf-mode*, and *default-sf-bits*. The default for flag is NIL. The bits parameter may be 8, 16, or 32. The values for the format and mode options are described below:

Format

snd-head-none
    The format is unknown and should be determined by reading the file.

dsnd-head-raw
    A raw format file has no header.

snd-head-AIFF
    AIFF format header.

snd-head-IRCAM
    IRCAM format header.

snd-head-NeXT
    1024-byte NeXT/SUN format header followed by IRCAM header ala CMIX. Note that the NeXT/SUN format has a header-length field, so it really is legal to have a large header, even though the normal minimal header is only 24 bytes. The additional space leaves room for maximum amplitudes, which...
can be used for normalizing floating-point soundfiles, and for other data. Nyquist follows the CMIX convention of placing an IRCAM format header immediately after the NeXT-style header.

\texttt{snd-head-Wave}  
Microsoft Wave format header.

\texttt{snd-head-flac}  
FLAC lossless compressed audio.

\texttt{snd-head-ogg}  
OGG-VORBIS compressed audio.

\texttt{snd-head-*}  
See \texttt{sndfnint.lsp} in the \texttt{nyquist/runtime} directory for more formats. The current list includes \texttt{paf, svx, nist, voc, w64, mat4, mat5, pvf, xi, htk, sds, avr, sd2, and caf}.

\textbf{Mode}

\texttt{snd-mode-adpcm}  
ADPCM mode (not supported).

\texttt{snd-mode-pcm}  
signed binary PCM mode.

\texttt{snd-mode-ulaw}  
8-bit U-Law mode.

\texttt{snd-mode-alaw}  
8-bit A-Law mode (not supported).

\texttt{snd-mode-float}  
32-bit floating point mode.

\texttt{snd-mode-upcm}  
unsigned binary PCM mode.

\texttt{snd-mode-*}  
See \texttt{sndfnint.lsp} in the \texttt{nyquist/runtime} for more modes. The current list includes \texttt{double, gsm610, dwv, dpcm, and msadpcm}.

The defaults for format, mode, and bits are as follows:
NeXT and Sun machines:

$snd\text{-}head\text{-}NeXT$, $snd\text{-}mode\text{-}pcm$, 16

SGI and Macintosh machines:

$snd\text{-}head\text{-}AIFF$, $snd\text{-}mode\text{-}pcm$, 16

\[s\text{-}read(filename, time\text{-}offset: offset, srate: sr, dur: dur, nchans: chans,\]

\[format: format, mode: mode, bits: n, swap: flag)[SAL]\]

\[(s\text{-}read filename :time\text{-}offset offset :srate sr :dur dur :nchans chans

:format format :mode mode :bits n :swap flag)[LISP]\]

Reads a sound from \textit{filename}. The global \texttt{*default-sf-dir*} applies. If a header is detected, the header is used to determine the format of the file, and header information overrides format information provided by keywords (except for \texttt{time\text{-}offset:} and \texttt{dur:}).

\[s\text{-}read("mysound.snd", srate: 44100)\]

specifies a sample rate of 44100 hz, but if the file has a header specifying 22050 hz, the resulting sample rate will be 22050. The parameters are:

- \texttt{offset} – the amount of time (in seconds) to skip from the beginning of the file. The default is 0.0.
- \texttt{sr} – the sample rate of the samples in the file. Default is \texttt{*default-sf-srate*}, which is normally 44100.
- \texttt{dur} – the maximum duration in seconds to read. Default is 10000.
- \texttt{chans} – the number of channels to read. It is assumed that samples from each channel are interleaved. Default is 1.
- \texttt{format} – the header format. See \texttt{s\text{-}save} for details. Default is \texttt{*default-sf\text{-}format*}, although this parameter is currently ignored.
- \texttt{mode} – the sample representation, e.g. PCM or float. See \texttt{s\text{-}save} for details. Default is \texttt{*default-sf\text{-}format*}.
- \texttt{n} – the number of bits per sample. See \texttt{s\text{-}save} for details. Default is \texttt{*default-sf\text{-}bits*}.
- \texttt{flag} – (T or NIL) swap byte order of each sample. Default is NIL.

If there is an error, for example if \texttt{offset} is greater than the length of the file, then NIL is returned rather than a sound. Information about the sound is also returned by \texttt{s\text{-}read} through \texttt{*rslt*}. The list assigned to \texttt{*rslt*} is of the form: (\textit{format channels mode bits swap samplerate duration flags byte\text{-}offset}), which are defined as follows:

- \texttt{format} – the header format. See \texttt{s\text{-}save} for details. Access this element of \texttt{*rslt*} by calling \texttt{snd\text{-}read\text{-}format(*rslt*)}.
- \texttt{channels} – the number of channels. Access this element of \texttt{*rslt*} by calling \texttt{snd\text{-}read\text{-}channels(*rslt*)}.

\footnote{Since XLISP does not support multiple value returns, multiple value returns are simulated by having the function assign additional return values in a list to the global variable \texttt{*rslt*}. Since this is a global, it should be inspected or copied immediately after the function return to insure that return values are not overwritten by another function.}
• **mode** – the sample representation, e.g. PCM or float. Access this element of *rslt* by calling `snd-read-mode(*rslt*)`. See `s-save` for details.

• **bits** – the number of bits per sample. Access this element of *rslt* by calling `snd-read-bits(*rslt*)`.

• **swap** – 1 if byte-swapping is needed, 0 otherwise. Access this element of *rslt* by calling `snd-read-swap(*rslt*)`.

• **samplerate** – the sample rate, expressed as a FLONUM. Access this element of *rslt* by calling `snd-read-srate(*rslt*)`.

• **duration** – the duration of the sound, in seconds. Access this element of *rslt* by calling `snd-read-dur(*rslt*)`.

• **flags** – The values for `format`, `channels`, `mode`, `bits`, `samplerate`, and `duration` are initially just the values passed in as parameters or default values to `s-read`. If a value is actually read from the sound file header, a flag is set. The flags are: `snd-head-type(format)`, `snd-head-channels`, `snd-head-mode`, `snd-head-bits`, `snd-head-srate`, and `snd-head-dur`. For example,

  ```lisp
  (let ((flags (s-read-flags *rslt*)))
    (not (zerop (logand flags snd-head-srate)))
  )
  ```

tells whether the sample rate was specified in the file. See also `sf-info` below.

• **byte-offset** – the byte offset into the file of the first sample to be read (this is used by the `s-overwrite` and `s-add-to` functions).

### s-add-to(expression, maxlen, filename [, offset]) [SAL]

**LISP**

Evaluates the expression, which should result in a sound or an array of sounds, and adds the result to the given filename. The global `*default-sf-dir*` applies. A FLONUM is returned, giving the maximum absolute value of all samples written. The sample rate(s) of expression must match those of the file. The maximum number of samples written per channel is given by `maxlen`, which allows writing the initial part of a very long or infinite sound. If `offset` is specified, the new sound is added to the file beginning at an offset from the beginning (in seconds). The file is extended if necessary to accommodate the new addition, but if `offset` falls outside of the original file, the file is not modified. (If necessary, use `s-add-to` to extend the file with zeros.) The file must be a recognized sound file with a header (not a raw sound file).

### s-overwrite(expression, maxlen, filename [, offset]) [SAL]

**LISP**

Evaluates the expression, which should result in a sound or an array of sounds, and replaces samples in the given filename. The global `*default-sf-dir*` applies. A FLONUM is returned, giving the maximum absolute value of all samples written. The sample rate(s) of expression must match those of the file. The maximum number of samples written per channel is given by `maxlen`, which allows writing the initial part of a very long or infinite sound. If `offset` is specified, the new sound is written to the file beginning at an offset from the beginning (in seconds). The file is extended if necessary to accommodate the new insert, but if `offset` falls outside of the original file, the file is not modified. (If necessary, use `s-add-to` to extend the file with zeros.) The file must be a recognized sound file with a header (not a raw sound file).
sf-info(filename) [SAL]
(sf-info filename) [LISP]
Prints information about a sound file. The parameter filename is a string. The file is assumed to be in *default-sf-dir* (see soundfilename below) unless the filename begins with "." or "/". The source for this function is in the runtime and provides an example of how to determine sound file parameters.

soundfilename(name) [SAL]
(soundfilename name) [LISP]
Converts a string name to a soundfile name. If name begins with "." or "/", the name is returned without alteration. Otherwise, a path taken from *default-sf-dir* is prepended to name. The s-plot, s-read, and s-save functions all use soundfilename to translate filenames.

s-plot(sound [, dur, n]) [SAL]
(s-plot sound [dur n]) [LISP]
Plots sound in a window. This function was designed to run a plot program on a Unix workstation, but now is primarily used with NyquistIDE, which has self-contained plotting. Normally, time/value pairs in ascii are written to points.dat and system-dependent code (or the NyquistIDE program) takes it from there. If the sound is longer than the optional dur (default is 2 seconds), only the first dur seconds are plotted. If there are more than n samples to be plotted, the signal is interpolated to have n samples before plotting. The data file used is *default-plot-file*:

*default-plot-file*
The file containing the data points, defaults to "points.dat".

spec-plot(sound [, offset, res: res, bw: bw, db: db]) [SAL]
(spec-plot sound [offset :res res :bw bw :db db]) [LISP]
Computes and plots the short-time magnitude spectrum of sound. The bandwidth of each bin is given by the res (resolution) keyword parameter, which defaults to *spec-plot-res*. Since one is often interested in lower frequencies, the bw (bandwidth keyword parameter limits the range of bins which are plotted, and defaults to *spec-plot-bw*. The output can be presented on a dB scale by setting the db keyword parameter to t (true). The defaults is to use a linear scale. The optional offset (in seconds) skips initial samples before taking a frame of samples from sound. The magnitude spectrum is sent to s-plot as a signal. The sample rate of the signal is set so that the plot labels on the horizontal axis represent Hz (not bin numbers.) To align bins with grid lines, one normally specifies res values in round numbers, e.g. 10 or 20. To achieve an arbitrary bin size, spec-plot resamples sound to a carefully computed sample rate that, after a power-of-2-sized FFT, yields the desired bin size.

*spec-plot-res*
The default bin resolution (separation in Hz between adjacent bins) for spec-plot. Defaults to 20 Hz. You can override this by using a keyword parameter when you call spec-plot, or for convenience, you can change this variable which will affect all future calls to spec-plot where the keyword parameter is omitted.
*spec-plot-bw*
The default highest frequency to include in `spec-plot`. Defaults to 8000 Hz. You can override this
by using a keyword parameter when you call `spec-plot`, or for convenience, you can change this
variable which will affect all future calls to `spec-plot` where the keyword parameter is omitted.

*spec-plot-db*
The default output from `spec-plot` displays magnitude on a linear scale, but there is an option to
display on a dB scale. You can change the default behavior by setting this variable to \( t \) (true), or you
can override the default in any call to `spec-plot` using a keyword parameter.

`s-print-tree(sound)` [SAL]
`(s-print-tree sound)` [LISP]
Prints an ascii representation of the internal data structures representing a sound. This is useful for
debugging Nyquist. Identical to `snd-print-tree`.

## 7.6 Low-level Functions

Nyquist includes many low-level functions that are used to implement the functions and behaviors described
in previous sections. For completeness, these functions are described here. Remember that these are low-level
functions that are not intended for normal use. Unless you are trying to understand the inner workings of
Nyquist, you can skip this section.

### 7.6.1 Creating Sounds

The basic operations that create sounds are described here.

`snd-const(value, t0, srate, duration)` [SAL]
`(snd-const value t0 srate duration)` [LISP]
Returns a sound with constant `value`, starting at `t0` with the given `duration`, at the sample rate `srate`.
You might want to use `pwl` (see Section 7.2.2.2) instead.

`snd-read(filename, offset, t0, format, channels, mode, bits, swap, sr, dur)` [SAL]
`(snd-read filename offset t0 format channels mode bits swap sr dur)` [LISP]
Loads a sound from a file with name `filename`. Files are assumed to consist of a header followed by
frames consisting of one sample from each channel. The `format` specifies the type of header, but this
information is currently ignored. Nyquist looks for a number of header formats and automatically
figures out which format to read. If a header can be identified, the header is first read from the file.
Then, the file pointer is advanced by the indicated `offset` (in seconds). If there is an unrecognized
header, Nyquist will assume the file has no header. If the header size is a multiple of the frame size
(bytes/sample * number-of-channels), you can use `offset` to skip over the header. To skip N bytes, use an `offset` of:
If the header is not a multiple of the frame size, either write a header or contact the author (dannenberg@cs.cmu.edu) for assistance. Nyquist will round offset to the nearest sample. The resulting sound will start at time \( t_0 \). If a header is found, the file will be interpreted according to the header information. If no header was found, \( \text{channels} \) tells how many channels there are, the samples are encoded according to \( \text{mode} \), the sample length is \( \text{bits} \), and \( \text{sr} \) is the sample rate. The \( \text{swap} \) flag is 0 or 1, where 1 means to swap sample bytes. The duration to be read (in seconds) is given by \( \text{dur} \). If \( \text{dur} \) is longer than the data in the file, then a shorter duration will be returned. If the file contains one channel, a sound is returned. If the file contains 2 or more channels, an array of sounds is returned. 

**Note:** you probably want to call \( \text{s-read} \) (see Section 7.5) instead of \( \text{snd-read} \). Also, see Section 7.5 for information on the \( \text{mode} \) and \( \text{format} \) parameters.

**snd-save**

\[ \text{snd-save(expression, maxlen, filename, format, mode, bits, swap, play)} \]

Evaluates the \( \text{expression} \), which should result in a sound or an array of sounds, and writes the result to the given \( \text{filename} \). If a multichannel sound (array) is written, the channels are up-sampled to the highest rate in any channel so that all channels have the same sample rate. The maximum number of samples written per channel is given by \( \text{maxlen} \), which allows writing the initial part of a very long or infinite sound. A header is written according to \( \text{format} \), samples are encoded according to \( \text{mode} \), using \( \text{bits} \) bits/sample, and swapping bytes if \( \text{swap} \) is 1 (otherwise it should be 0). If \( \text{play} \) is not null, the audio is played in real time (to the extent possible) as it is computed. The peak value of the sound is returned. In addition, the symbol \(*\text{RSLT}*\) is bound to a list containing the sample rate, number of channels, and duration (in that order) of the saved sound. 

**Note:** you probably want to call \( \text{s-save} \) (see Section 7.5) instead. The \( \text{format} \) and \( \text{mode} \) parameters are described in Section 7.5.

**snd-overwrite**

\[ \text{snd-overwrite(expression, maxlen, filename, offset, format, mode, bits, swap)} \]

Evaluates the \( \text{expression} \), which should result in a sound or an array of sounds, and replaces samples in the given \( \text{filename} \), writing the first frame at a time of \( \text{offset} \) seconds. The \( \text{offset} \) must be less than or equal to the duration of the existing file. The duration of the written samples may be greater than that of the file, in which case the file is extended as necessary. The sample rate(s) of \( \text{expression} \) and the number of channels must match those of the file. If \( \text{format} \) is \text{SND-HEAD-RAW}, then the file format is given by \( \text{mode} \) (see \( \text{snd-save} \), \( \text{bits} \) (per channel), \( \text{swap} \) (1 means to swap bytes and 0 means write them in the native byte order), and the number of channels and sample rate of the sound returned by evaluating \( \text{expression} \). If the file is a known audio file format, \( \text{format} \) should be \text{SND-HEAD-NONE}, and the other parameters are ignored. Up to a maximum of \( \text{maxlen} \) samples will be written per channel. The peak value of the sound is returned. In addition, the symbol \(*\text{RSLT}*\) is bound to a list containing the duration of the written sound (which may not be the duration of the sound file). Use \( \text{s-add-to} \) (in Section 7.5 or \( \text{s-overwrite} \) (in Section 7.5 instead of this function).
snd-coterm\((s1, s2)\) [SAL]
\((\text{snd-coterm } s1 \ s2)\) [LISP]
Returns a copy of \(s1\), except the start time is the maximum of the start times of \(s1\) and \(s2\), and the termination time is the minimum of \(s1\) and \(s2\). (After the termination time, the sound is zero as if \(s1\) is gated by \(s2\).) Some rationale follows: In order to implement \(s\)-add-to, we need to read from the target sound file, add the sounds to a new sound, and overwrite the result back into the file. We only want to write as many samples into the file as there are samples in the new sound. However, if we are adding in samples read from the file, the result of a \(s\)-add in Nyquist will have the maximum duration of either sound. Therefore, we may read to the end of the file. What we need is a way to truncate the read, but we cannot easily do that, because we do not know in advance how long the new sound will be. The solution is to use \(\text{snd-coterm}\), which will allow us to truncate the sound that is read from the file (\(s1\)) according to the duration of the new sound (\(s2\)). When this truncated sound is added to the new sound, the result will have only the duration of the new sound, and this can be used to overwrite the file. This function is used in the implementation of \(s\)-add-to, which is defined in runtime/fileio.lsp.

\((\text{snd-from-array } \ldots)\) [SAL]
\((\text{snd-from-array } \ldots)\) [LISP]
See page 68.

snd-white\((t0, sr, d)\) [SAL]
\((\text{snd-white } t0 \ sr \ d)\) [LISP]
Generate white noise, starting at \(t0\), with sample rate \(sr\), and duration \(d\). You probably want to use \(\text{noise}\) (see Section 7.2.2.7).

snd-zero\((t0, sr)\) [SAL]
\((\text{snd-zero } t0 \ sr)\) [LISP]
Creates a sound that is zero everywhere, starts at \(t0\), and has sample rate \(sr\). The logical stop time is immediate, i.e. also at \(t0\). You probably want to use \(\text{pwl}\) (see Section 7.2.2.2) instead.

## 7.6.2 Signal Operations

This next set of functions take sounds as arguments, operate on them, and return a sound.

snd-abs\((sound)\) [SAL]
\((\text{snd-abs } sound)\) [LISP]
Computes a new sound where each sample is the absolute value of the corresponding sample in \(sound\). You should probably use \(\text{s-\text{abs}}\) instead. (See Section 7.2.2.7.)

snd-sqrt\((sound)\) [SAL]
\((\text{snd-sqrt } sound)\) [LISP]
Computes a new sound where each sample is the square root of the corresponding sample in \(sound\). If a sample is negative, it is taken to be zero to avoid raising a floating point error. You should probably use \(\text{s-sqrt}\) instead. (See Section 7.2.2.7.)
`snd-add(sound1, sound)` [SAL]

`(snd-add sound1 sound)` [LISP]

Adds two sounds. The resulting start time is the minimum of the two parameter start times, the logical stop time is the maximum of the two parameter stop times, and the sample rate is the maximum of the two parameter sample rates. Use `sim` or `sum` instead of `snd-add` (see Section 7.4).

`snd-offset(sound, offset)` [SAL]

`(snd-offset sound offset)` [LISP]

Add an offset to a sound. The resulting start time, logical stop time, stop time, and sample rate are those of `sound`. Use `sum` instead (see Section 7.4).

`snd-avg(sound, blocksize, stepsize, operation)` [SAL]

`(snd-avg sound blocksize stepsize operation)` [LISP]

Computes the averages or peak values of blocks of samples. Each output sample is an average or peak of `blocksize` (a fixnum) adjacent samples from the input `sound`. After each average or peak is taken, the input is advanced by `stepsize`, a fixnum which may be greater or less than `blocksize`. The output sample rate is the `sound` (input) sample rate divided by `stepsize`. The duration of the output is the same (approximately, due to rounding) as that of `sound`. Notice however, that the features of the input will appear earlier in the output by half the window size. For example, a sharp peak in the input will result in a smoothed peak (using `OP-AVERAGE`) one half `blocksize` earlier. You can correct for this shift by inserting one half `blocksize` of silence before `sound`, e.g. if `s` has a sample rate of 44100 Hz, then `snd-avg(seq(s-rest(0.01), cue(s)), 882, 441, OP-AVERAGE)` will shift `s` by 0.01 s to compensate for the shift introduced by a smoothing window of size 0.02 s (882/44100). This function is useful for computing low-sample-rate rms or peak amplitude signals for input to `snd-gate` or `snd-follow`. To select the operation, `operation` should be one of `OP-AVERAGE` or `OP-PEAK`. (These are global lisp variables; the actual `operation` parameter is an integer.) For RMS computation, see `rms` in Section 7.2.2.7.

`snd-clip(sound, peak)` [SAL]

`(snd-clip sound peak)` [LISP]

Hard limit `sound` to the given `peak`, a positive number. The samples of `sound` are constrained between an upper value of `peak` and a lower value of `-peak`. Use `clip` instead (see Section 7.2.2.7).

`snd-compose(f, g)` [SAL]

`(snd-compose f g)` [LISP]

Compose two signals, i.e. compute \( f(g(t)) \), where \( f \) and \( g \) are sounds. This function is used primarily to implement time warping, but it can be used in other applications such as frequency modulation. For each sample \( x \) in \( g \), `snd-compose` looks up the value of \( f(x) \) using linear interpolation. The resulting sample rate, start time, etc. are taken from \( g \). The sound \( f \) is used in effect as a lookup table, but it is assumed that \( g \) is non-decreasing, so that \( f \) is accessed in time order. This allows samples of \( f \) to be computed and discarded incrementally. If in fact \( g \) decreases, the current sample of \( g \) is replaced by the previous one, forcing \( g \) into compliance with the non-decreasing restriction. See also `sref`, `shape`, and `snd-resample`.

For an extended example that uses `snd-compose` for variable pitch shifting, see demos/pitch_change.htm.
A variable delay: \textit{sound} is delayed by the sum of \textit{offset} (a \textsc{FixNum} or \textsc{FLoNum}) and \textit{vardelay} (a \textsc{Sound}). The specified delay is adjusted to lie in the range of zero to \textit{maxdelay} seconds to yield the actual delay, and the delay is implemented using linear interpolation. This function was designed specifically for use in a chorus effect: the \textit{offset} is set to half of \textit{maxdelay}, and the \textit{vardelay} input is a slow sinusoid. The maximum delay is limited to \textit{maxdelay}, which determines the length of a fixed-sized buffer. The function \texttt{tapv} is equivalent and preferred (see Section 7.2.2.3).

A variable delay like \texttt{snd-tapv} except there is no linear interpolation. By eliminating interpolation, the output is an exact copy of the input with no filtering or distortion. On the other hand, delays jump by samples causing samples to double or skip even when the delay is changed smoothly.

Makes a copy of \textit{sound}. Since operators always make (logical) copies of their sound parameters, this function should never be needed. This function is here for debugging.

Linear interpolation of samples down to the given sample rate \textit{srate}, which must be lower than the sample rate of \textit{sound}. Do not call this function. Nyquist performs sample-rate conversion automatically as needed. If you want to force a conversion, call \texttt{force-srate} (see Section 7.2.2).

Compute the exponential of each sample of \textit{sound}. Use \texttt{s-exp} instead (see Section 7.2.2.7).

An envelope follower. The basic goal of this function is to generate a smooth signal that rides on the peaks of the input signal. The usual objective is to produce an amplitude envelope given a low-sample rate (control rate) signal representing local RMS measurements. The first argument is the input signal. The \textit{floor} is the minimum output value. The \textit{risetime} is the time (in seconds) it takes for the output to rise (exponentially) from \textit{floor} to unity (1.0) and the \textit{falltime} is the time it takes for the output to fall (exponentially) from unity to \textit{floor}. The algorithm looks ahead for peaks and will begin to increase the output signal according to \textit{risetime} in anticipation of a peak. The amount of anticipation (in samples) is given by \textit{lookahead}. The algorithm is as follows: the output value is allowed to increase according to \textit{risetime} or decrease according to \textit{falltime}. If the next input sample is in this range, that sample is simply output as the next output sample. If the next input sample is too large, the algorithm goes back in time as far as necessary to compute an envelope that rises according to \textit{risetime} to meet the new value. The algorithm will only work backward as far as \textit{lookahead}. If that is not far enough, then there is a final forward pass computing a rising signal from the earliest output sample. In this case, the output
signal will be at least momentarily less than the input signal and will continue to rise exponentially until it intersects the input signal. If the input signal falls faster than indicated by \textit{falltime}, the output fall rate will be limited by \textit{falltime}, and the fall in output will stop when the output reaches \textit{floor}. This algorithm can make two passes through the buffer on sharply rising inputs, so it is not particularly fast. With short buffers and low sample rates this should not matter. See \texttt{snd-avg} above for a function that can help to generate a low-sample-rate input for \texttt{snd-follow}. See \texttt{snd-chase} in Section 7.6.3 for a related filter.

\texttt{snd-gate(sound, lookahead, risetime, falltime, floor, threshold)} [SAL]
\texttt{(snd-gate sound lookahead risetime falltime floor threshold)} [LISP]

This function generates an exponential rise and decay intended for noise gate implementation. The decay starts when the signal drops below threshold and stays there for longer than lookahead. Decay continues until the value reaches floor, at which point the decay stops and the output value is held constant. Either during the decay or after the floor is reached, if the signal goes above threshold, then the output value will rise to unity (1.0) at the point the signal crosses the threshold. Again, look-ahead is used, so the rise actually starts before the signal crosses the threshold. The rise is a constant-rate exponential and set so that a rise from \textit{floor} to unity occurs in \textit{risetime}. Similarly, the fall is a constant-rate exponential such that a fall from unity to \textit{floor} takes \textit{falltime}. The result is delayed by \textit{lookahead}, so the output is not actually synchronized with the input. To compensate, you should drop the initial \textit{lookahead} of samples. Thus, \texttt{snd-gate} is not recommended for direct use. Use \texttt{gate} instead (see Section 7.1.4).

\texttt{snd-inverse(signal, start, srate)} [SAL]
\texttt{(snd-inverse signal start srate)} [LISP]

Compute the function inverse of \texttt{signal}, that is, compute \( g(t) \) such that \( signal(g(t)) = t \). This function assumes that \texttt{signal} is non-decreasing, it uses linear interpolation, the resulting sample rate is \texttt{srate}, and the result is shifted to have a starting time of \texttt{start}. If \texttt{signal} decreases, the true inverse may be undefined, so we define \texttt{snd-inverse} operationally as follows: for each output time point \( t \), scan ahead in \texttt{signal} until the value of \texttt{signal} exceeds \( t \). Interpolate to find an exact time point \( x \) from \texttt{signal} and output \( x \) at time \( t \). This function is intended for internal system use in implementing time warps.

\texttt{snd-log(sound)} [SAL]
\texttt{(snd-log sound)} [LISP]

Compute the natural logarithm of each sample of \texttt{sound}. Use \texttt{s-log} instead (see Section 7.2.2.7).

\texttt{peak(expression, maxlen)} [SAL]
\texttt{(peak expression maxlen)} [LISP]

Compute the maximum absolute value of the amplitude of a sound. The sound is created by evaluating \texttt{expression} (as in \texttt{s-save}). Only the first \texttt{maxlen} samples are evaluated. The \texttt{expression} is automatically quoted (peak is a macro), so do not quote this parameter. If \texttt{expression} is a variable, then the \texttt{global binding} of that variable will be used. Also, since the variable retains a reference to the sound, the sound will be evaluated and left in memory. See Section 5.3 on page 43 for examples.
Compute the maximum absolute value of the amplitude of a sound. The sound is created by evaluating
expression (as in snd-save), which is therefore normally quoted by the caller. At most maxlen samples are computed. The result is the maximum of the absolute values of the samples. Notes: It is recommended to use peak (see above) instead. If you want to find the maximum of a sound bound to a local variable and it is acceptable to save the samples in memory, then this is probably the function to call. Otherwise, use peak.

snd-maxv(sound1, sound2) [SAL]
(snd-maxv sound1 sound2) [LISP]
Compute the maximum of sound1 and sound2 on a sample-by-sample basis. The resulting sound has its start time at the maximum of the input start times and a logical stop at the minimum logical stop of the inputs. The physical stop time is the minimum of the physical stop times of the two sounds. Note that this violates the “normal” interpretation that sounds are zero outside their start and stop times. For example, even if sound1 extends beyond sound2 and is greater than zero, the result value in this extension will be zero because it will be after the physical stop time, whereas if we simply treated sound2 as zero in this region and took the maximum, we would get a non-zero result. Use a-max instead (see Section 7.2.2.7).

snd-normalize(sound) [SAL]
(snd-normalize sound) [LISP]
Internally, sounds are stored with a scale factor that applies to all samples of the sound. All operators that take sound arguments take this scale factor into account (although it is not always necessary to perform an actual multiply per sample), so you should never need to call this function. This function multiplies each sample of a sound by its scale factor, returning a sound that represents the same signal, but whose scale factor is 1.0.

snd-oneshot(sound, threshold, ontime) [SAL]
(snd-oneshot sound threshold ontime) [LISP]
Computes a new sound that is zero except where sound exceeds threshold. From these points, the result is 1.0 until sound remains below threshold for ontime (in seconds). The result has the same sample rate, start time, logical stop time, and duration as sound.

snd-prod(sound1, sound2) [SAL]
(snd-prod sound1 sound2) [LISP]
Computes the product of sound1 and sound2. The resulting sound has its start time at the maximum of the input start times and a logical stop at the minimum logical stop of the inputs. Do not use this function. Use mult or prod instead (see Section 7.2.2). Sample rate, start time, etc. are taken from sound.

snd-pwl(t0, sr, lis) [SAL]
(snd-pwl t0 sr lis) [LISP]
Computes a piece-wise linear function according to the breakpoints in lis. The starting time is t0, and the sample rate is sr. The breakpoints are passed in an XLISP list (of type LVAL) where the list
alternates sample numbers (\texttt{FIXNUMs}, computed in samples from the beginning of the \texttt{pwl} function) and values (the value of the \texttt{pwl} function, given as a \texttt{FLONUM}). There is an implicit starting point of \((0, 0)\). The list must contain an odd number of points, the omitted last value being implicitly zero \((0)\). The list is assumed to be well-formed. Do not call this function. Use \texttt{pwl} instead (see Section 7.2.2.2).

\textbf{snd-quantize} \((\texttt{sound}, \texttt{steps})\) [SAL]
\texttt{(snd-quantize sound steps)} [LISP]
Quantizes a sound. See Section 7.2.2.7 for details.

\textbf{snd-recip} \((\texttt{sound})\) [SAL]
\texttt{(snd-recip sound)} [LISP]
Compute the reciprocal of each sample of \texttt{sound}. Use \texttt{recip} instead (see Section 7.2.2.7).

\textbf{snd-resample} \((\texttt{f}, \texttt{rate})\) [SAL]
\texttt{(snd-resample f rate)} [LISP]
Resample \texttt{sound} \(f\) using high-quality interpolation, yielding a new sound with the specified \texttt{rate}. The result is scaled by \(0.95\) because often, in resampling, interpolated values exceed the original sample values, and this could lead to clipping. The resulting start time, etc. are taken from \(f\). Use \texttt{resample} instead.

\textbf{snd-resamplev} \((\texttt{f}, \texttt{rate}, \texttt{g})\) [SAL]
\texttt{(snd-resamplev f rate g)} [LISP]
Compose two signals, i.e. compute \(f(g(t))\), where \(f\) and \(g\) are sounds. The result has sample rate given by \texttt{rate}. At each time \(t\) (according to the \texttt{rate}), \(g\) is linearly interpolated to yield an increasing sequence of high-precision score-time values. \(f\) is then interpolated at each value to yield a result sample. If in fact \(g\) decreases, the current sample of \(g\) is replaced by the previous one, forcing \(g\) into compliance with the non-decreasing restriction. The result is scaled by \(0.95\) because often, in resampling, interpolated values exceed the original sample values, and this could lead to clipping. Note that if \(g\) has a high sample rate, this may introduce unwanted jitter into sample times. See \texttt{sound-warp} for a detailed discussion. See \texttt{snd-compose} for a fast, low-quality alternative to this function. Normally, you should use \texttt{sound-warp} instead of this function.

\textbf{snd-scale} \((\texttt{scale}, \texttt{sound})\) [SAL]
\texttt{(snd-scale scale sound)} [LISP]
Scales the amplitude of \texttt{sound} by the factor \texttt{scale}. Use \texttt{scale} instead (see Section 7.2.2).

\textbf{snd-shape} \((\texttt{signal}, \texttt{table}, \texttt{origin})\) [SAL]
\texttt{(snd-shape signal table origin)} [LISP]
A waveshaping function. This is the primitive upon which \texttt{shape} is based. The \texttt{snd-shape} function is like \texttt{shape} except that \texttt{signal} and \texttt{table} must be (single-channel) sounds. Use \texttt{shape} instead (see Section 7.2.2.3).

\textbf{snd-up} \((\texttt{srate}, \texttt{sound})\) [SAL]
\texttt{(snd-up srate sound)} [LISP]
Increases sample rate by linear interpolation. The \texttt{sound} is the signal to be up-sampled, and \texttt{srate} is the output sample rate. Do not call this function. Nyquist performs sample-rate conversion automatically as needed. If you want to force a conversion, call \texttt{force-srate} (see Section 7.2.2).
Nyquist Reference Manual

Chapter 7. Nyquist Functions

nyquist Reference Manual

Chapter 7. Nyquist Functions

snd-xform(sound, sr, time, start, stop, scale) [SAL]

(snd-xform sound sr time start stop scale) [LISP]

Makes a copy of sound and then alters it in the following order: (1) the start time (snd-t0) of the sound is shifted to time, (1) the sound is stretched as a result of setting the sample rate to sr (the start time is unchanged by this), (3) the sound is clipped from start to stop, (4) if start is greater than time, the sound is shifted shifted by time - start, so that the start time is time, (5) the sound is scaled by scale. An empty (zero) sound at time will be returned if all samples are clipped. Normally, you should accomplish all this using transformations. A transformation applied to a sound has no effect, so use cue to create a transformable sound (see Section 7.2.1).

snd-yin(sound, minstep, maxstep, rate) [SAL]

(snd-yin sound minstep maxstep rate) [LISP]

Identical to yin. See Section 7.2.2.7.

7.6.3 Filters

These are also "Signal Operators," the subject of the previous section, but there are so many filter functions, they are documented in this special section.

Some filters allow time-varying filter parameters. In these functions, filter coefficients are calculated at the sample rate of the filter parameter, and coefficients are not interpolated.

snd-alpass(sound, delay, feedback) [SAL]

(snd-alpass sound delay feedback) [LISP]

An all-pass filter. This produces a repeating echo effect without the resonances of snd-delay. The feedback should be less than one to avoid exponential amplitude blowup. Delay is rounded to the nearest sample. You should use alpass instead (see Section 7.2.2.3).

snd-alpasscv(sound, delay, feedback) [SAL]

(snd-alpasscv sound delay feedback) [LISP]

An all-pass filter with variable feedback. This is just like snd-alpass except feedback is a sound. You should use alpass instead (see Section 7.2.2.3).

snd-alpassvv(sound, delay, feedback, maxdelay) [SAL]

(snd-alpassvv sound delay feedback maxdelay) [LISP]

An all-pass filter with variable feedback and delay. This is just like snd-alpass except feedback and delay are sounds, and there is an additional FLONUM parameter, maxdelay, that gives an upper bound on the value of delay. Note: delay must remain between zero and maxdelay. If not, results are undefined, and Nyquist may crash. You should use alpass instead (see Section 7.2.2.3).

snd-areson(sound, hz, bw, normalization) [SAL]

(snd-areson sound hz bw normalization) [LISP]

A notch filter modeled after the areson unit generator in Csound. The snd-areson filter is an exact complement of snd-reson such that if both are applied to the same signal with the same parameters,
the sum of the results yields the original signal. Note that because of this complementary design, the power is not normalized as in snd-reson. See snd-reson for details on normalization. You should use areson instead (see Section 7.2.2.3).

snd-aresoncv(sound, hz, bw, normalization) [SAL]
 snd-aresoncv sound hz bw normalization) [LISP]
 This function is identical to snd-areson except the bw (bandwidth) parameter is a sound. Filter coefficients are updated at the sample rate of bw. The "cv" suffix stands for Constant, Variable, indicating that hz and bw are constant (a number) and variable (a sound), respectively. This naming convention is used throughout. You should use areson instead (see Section 7.2.2.3).

snd-aresonvc(sound, hz, bw, normalization) [SAL]
 (snd-aresonvc sound hz bw normalization) [LISP]
 This function is identical to snd-areson except the hz (center frequency) parameter is a sound. Filter coefficients are updated at the sample rate of hz. You should use areson instead (see Section 7.2.2.3).

snd-aresonvv(sound, hz, bw, normalization) [SAL]
 (snd-aresonvv sound hz bw normalization) [LISP]
 This function is identical to snd-areson except both hz (center frequency) and bw (bandwidth) are sounds. Filter coefficients are updated at the next sample of either hz or bw. You should use areson instead (see Section 7.2.2.3).

snd-atone(sound, hz) [SAL]
 (snd-atone sound hz) [LISP]
 A high-pass filter modeled after the atone unit generator in Csound. The snd-atone filter is an exact complement of snd-tone such that if both are applied to the same signal with the same parameters, the sum of the results yields the original signal. You should use hp instead (see Section 7.2.2.3).

snd-atonev(sound, hz) [SAL]
 (snd-atonev sound hz) [LISP]
 This is just like snd-atone except that the hz cutoff frequency is a sound. Filter coefficients are updated at the sample rate of hz. You should use hp instead (see Section 7.2.2.3).

snd-biquad(sound, b0, b1, b2, a1, a2, z1init, z2init) [SAL]
 (snd-biquad sound b0 b1 b2 a1 a2 z1init z2init) [LISP]
 A general second order IIR filter, where a0 is assumed to be unity. For a1 and a2, the sign convention is opposite to that of Matlab. All parameters except the input sound are of type FLONUM. You should probably use one of lowpass2, highpass2, bandpass2, notch2, allpass2, eq-lowshelf, eq-highshelf, eq-band, lowpass4, lowpass6, lowpass8, highpass4, highpass6, or highpass8, which are all based on snd-biquad and described in Section 7.2.2.3. For completeness, you will also find biquad and biquad-m described in that section.

snd-chase(sound, risetime, falltime) [SAL]
 (snd-chase sound risetime falltime) [LISP]
 A slew rate limiter. The output "chases" the input at rates determined by risetime and falltime. If the input changes too fast, the output will lag behind the input. This is a form of lowpass filter, but it was
created to turn hard-switching square waves into smoother control signals that could be used for linear
crossfades. If the input switches from 0 to 1, the output will linearly rise to 1 in \textit{risetime} seconds. If the
input switches from 1 to 0, the output will linearly fall to 0 in \textit{falltime} seconds. The generated slope is
constant; the transition is linear; this is not an exponential rise or fall. The \textit{risetime} and \textit{falltime} must
be scalar constants; complain to the author if this is not adequate. The \texttt{snd-chase} function is safe for
ordinary use. See \texttt{snd-follow} in Section 7.6.2 for a related function.

\begin{verbatim}
snd-congen(\texttt{gate, risetime, falltime}) [SAL]
(Snd-congen \texttt{gate risetime falltime}) [LISP]
\end{verbatim}

A simple "contour generator" based on analog synthesizers. The \texttt{gate} is a sound that normally steps
from 0.0 to 1.0 at the start of an envelop and goes from 1.0 back to 0.0 at the beginning of the release.
At each sample, the output converges to the input exponentially. If \texttt{gate} is greater than the output, e.g.
the attack, then the output converges half-way to the output in \textit{risetime}. If the \texttt{gate} is less than the
output, the half-time is \textit{falltime}. The sample rate, starting time, logical-stop-time, and terminate time
are taken from \texttt{gate}. You should use \texttt{congen} instead (see Section 7.2.2.3).

\begin{verbatim}
snd-convolve(\texttt{sound, response}) [SAL]
(Snd-convolve \texttt{sound response}) [LISP]
\end{verbatim}

Convolves \texttt{sound} by \texttt{response} using a fast convolution algorithm. The \texttt{sound} can be any length, but the
\texttt{response} is computed and stored in in memory. The required computation time per sample and total
space are proportional to the length of \texttt{response}. Use \texttt{convolve} instead (see Section 7.2.2.3).

\begin{verbatim}
snd-delay(\texttt{sound, delay, feedback}) [SAL]
(Snd-delay \texttt{sound delay feedback}) [LISP]
\end{verbatim}

Feedback delay. The output, initially \texttt{sound}, is recursively delayed by \texttt{delay}, scaled by \texttt{feedback},
and added to itself, producing an repeating echo effect. The \texttt{feedback} should be less than one to
avoid exponential amplitude blowup. Delay is rounded to the nearest sample. You should use
\texttt{feedback-delay} instead (see Section 7.2.2.3).

\begin{verbatim}
snd-delaycv(\texttt{sound, delay, feedback}) [SAL]
(Snd-delaycv \texttt{sound delay feedback}) [LISP]
\end{verbatim}

Feedback delay with variable \texttt{feedback}. This is just like \texttt{snd-delay} except \texttt{feedback} is a sound. You
should use \texttt{feedback-delay} instead (see Section 7.2.2.3).

\begin{verbatim}
snd-reson(\texttt{sound, hz, bw, normalization}) [SAL]
(Snd-reson \texttt{sound hz bw normalization}) [LISP]
\end{verbatim}

A second-order resonating (bandpass) filter with center frequency \texttt{hz} and bandwidth \texttt{bw}, modeled
after the \texttt{reson} unit generator in Csound. The \texttt{normalization} parameter must be an integer and (like
in Csound) specifies a scaling factor. A value of 1 specifies a peak amplitude response of 1.0; all
frequencies other than \texttt{hz} are attenuated. A value of 2 specifies the overall RMS value of the amplitude
response is 1.0; thus filtered white noise would retain the same power. A value of zero specifies no
scaling. The result sample rate, start time, etc. are takend from \texttt{sound}. You should use \texttt{reson} instead
(see Section 7.2.2.3).
This function is identical to `snd-reson` except `bw` (bandwidth) is a sound. Filter coefficients are updated at the sample rate of `bw`. You should use `reson` instead (see Section 7.2.2.3).

This function is identical to `snd-reson` except `hz` (center frequency) is a sound. Filter coefficients are updated at the sample rate of `hz`. You should use `reson` instead (see Section 7.2.2.3).

This function is identical to `snd-reson` except both `hz` (center frequency) and `bw` (bandwidth) are sounds. Filter coefficients are updated at the next sample from either `hz` or `bw`. You should use `reson` instead (see Section 7.2.2.3).

A chorus implemented in STK. The parameter `delay` is a FIXNUM representing the median desired delay length in samples. A typical value is 6000. The FLONUM parameters `depth` and `freq` set the modulation depth (from 0 to 1) and modulation frequency (in Hz), `mix` sets the mixture of input sound and chorused sound, where a value of 0.0 means input sound only (dry) and a value of 1.0 means chorused sound only (wet). You should use `stkchorus` instead (see Section 7.2.2.4).

A pitch shifter implemented in STK. The `sound` is shifted in pitch by `shift`, a FLONUM representing the shift factor. A value of 1.0 means no shift. The parameter `mix` sets the mixture of input and shifted sounds. A value of 0.0 means input only (dry) and a value of 1.0 means shifted sound only (wet). You should use `pitshift` instead (see Section 7.2.2.4).

A reverb implemented in STK. The parameter `rev-type` is a FIXNUM ranging from zero to two and selects the type of reverb. Zero selects NRev type, one selects JCrev, and two selects PRCRev. The input `sound` is processed by the reverb with a decay time in seconds (a FLONUM). The `mix`, a FLONUM, sets the mixture of dry input and reverb output. A value of 0.0 means input only (dry) and a value of 1.0 means reverb only (wet). The sample rate is that of `sound`. You should use `nrev`, `jcrev` or `prcrev` instead (see Section 7.2.2.4).

A first-order recursive low-pass filter, based on the `tone` unit generator of Csound. The `hz` parameter is the cutoff frequency, the response curve’s half-power point. The result sample rate, start time, etc. are taken from `sound`. You should use `lp` instead (see Section 7.2.2.3).
**Nyquist Functions**

**Table-Lookup Oscillator Functions**

These functions all use a sound to describe one period of a periodic waveform. In the current implementation, the sound samples are copied to an array (the waveform table) when the function is called. To make a table-lookup oscillator generate a specific pitch, we need to have several pieces of information:

- A waveform to put into the table. This comes from the `sound` parameter.
- The length (in samples) of the waveform. This is obtained by reading samples (starting at the sound’s start time, not necessarily at time zero) until the physical stop time of the sound. (If you read the waveform from a file or generate it with functions like `sim` and `sine`, then the physical and logical stop times will be the same and will correspond to the duration you specified, rounded to the nearest sample.)
- The intrinsic sample rate of the waveform. This sample rate is simply the sample rate property of `sound`.
- The pitch of the waveform. This is supplied by the `step` parameter and indicates the pitch (in steps) of `sound`. You might expect that the pitch would be related to the period (length) of `sound`, but there is the interesting case that synthesis based on sampling often loops over multiple periods. This means that the fundamental frequency of a generated tone may be some multiple of the looping rate. In Nyquist, you always specify the perceived pitch of the looped `sound` if the sound is played at the `sound`’s own sample rate.
- The desired pitch. This is specified by the `hz` parameter in Hertz (cycles per second) in these low-level functions. Note that this is not necessarily the "loop" rate at which the table is scanned. Instead, Nyquist figures what sample rate conversion would be necessary to "transpose" from the `step` which specifies the original pitch of `sound` to `hz`, which gives the desired pitch. The mixed use of steps and Hertz came about because it seemed that sample tables would be tagged with steps ("I sampled a middle-C"), whereas frequency deviation in the `fmosc` function is linear, thus calling for a specification in Hertz.
- The desired sample rate. This is given by the `sr` parameter in Hertz.

Other parameters common to all of these oscillator functions are:

- `t0`, the starting time, and
- `phase`, the starting phase in degrees. Note that if the `step` parameter indicates that the table holds more than one fundamental period, then a starting phase of 360 will be different than a starting phase of 0.
snd-amosc(sound, step, sr, hz, t0, am, phase) [SAL]

(snd-amosc sound step sr hz t0 am phase) [LISP]
An oscillator with amplitude modulation. The sound am specifies the amplitude and the logical stop time. The physical stop time is also that of am. You should use amosc instead (see Section 7.2.2.1).

snd-fmosc(s, step, sr, hz, t0, fm, phase) [SAL]

(snd-fmosc s step sr hz t0 fm phase) [LISP]
A Frequency Modulation oscillator. The sound fm specifies frequency deviation (in Hertz) from hz. You should use fmosc instead (see Section 7.2.2.1).

snd-fmfb(t0, hz, sr, index, dur) [SAL]

(snd-fmfb t0 hz sr index dur) [LISP]
A Feedback FM oscillator. The resulting sound starts at t0, has a fundamental frequency of hz, a sample rate of sr, and a duration of dur seconds. The index is a FLONUM that specifies the amount of feedback. You should use fmfb instead (see Section 7.2.2.1).

snd-fmfv(t0, hz, sr, index)

(snd-fmfv t0 hz sr index) [LISP]
A Feedback FM oscillator. The resulting sound starts at t0, has a fundamental frequency of hz, and a sample rate of sr. The index is a SOUND that specifies the amount of feedback and determines the duration. You should use fmfb instead (see Section 7.2.2.1).

snd-buzz(n, sr, hz, t0, fm) [SAL]

(snd-buzz n sr hz t0 fm) [LISP]
A buzz oscillator, which generates n harmonics of equal amplitude. The fm specifies frequency deviation (in Hertz) from hz. You should use buzz instead (see Section 7.2.2.1).

snd-pluck(sr, hz, t0, d, final-amp) [SAL]

(snd-pluck sr hz t0 d final-amp) [LISP]
A Karplus-Strong plucked string oscillator with sample rate sr, fundamental frequency hz, starting time t0, duration d, initial amplitude approximately 1.0 (not exact because the string is initialized with random values) and final amplitude approximately final-amp. You should use pluck instead (see Section 7.2.2.1).

snd-osc(s, step, sr, hz, t0, d, phase) [SAL]

(snd-osc s step sr hz t0 d phase) [LISP]
A simple table lookup oscillator with fixed frequency. The duration is d seconds. You should use osc instead (see Section 7.2.2.1).

snd-partial(sr, hz, t0, env) [SAL]

(snd-partial sr hz t0 env) [LISP]
This is a special case of snd-amosc that generates a sinusoid starting at phase 0 degrees. The env parameter gives the envelope or any other amplitude modulation. You should use partial instead (see Section 7.2.2.1).
snd-sine(t0, hz, sr, d) [SAL]
nsnd-sine t0 hz sr d) [LISP]

This is a special case of snd-osc that always generates a sinusoid with initial phase of 0 degrees. You should use sine instead (see Section 7.2.2.1).

snd-sampler(s, step, start, sr, hz, t0, fm, npoints) [SAL]
nsnd-sampler s step start sr hz t0 fm npoints) [LISP]

Returns a sound constructed by reading a sample from beginning to end and then splicing on copies of the same sound from a loop point to the end. The sound s is the source sound to be looped, and step (a FLONUM) is the nominal fundamental frequency (in steps, not Hz) of s. The start (a FLONUM) is the time in seconds at which to start the loop, sr (a FLONUM) is the desired sample rate of the output, hz is the nominal fundamental frequency of the output, t0 (a FLONUM) is the starting time of the output, and fm (a SOUND) is frequency modulation that is added to hz to determine the output fundamental frequency. The parameter npoints (a FIXNUM) specifies how many points should be used for sample interpolation. Currently this parameter defaults to 2 and only 2-point (linear) interpolation is implemented. It is an error to modulate such that the frequency is negative. Note also that the loop point may be fractional. This function implements a typical sampling synthesis algorithm, looping and resampling the input according to the ratio between the desired fundamental frequency (which is the sum of hz and fm) and the nominal fundamental of the looped sound (which is assumed to be given by step). You should use sampler instead (see Section 7.2.2.1).

snd-siosc(tables, sr, hz, t0, fm) [SAL]
nsnd-siosc tables sr hz t0 fm) [LISP]

A Spectral Interpolation Oscillator with frequency modulation. The tables is a list of sounds and sample counts as follows: (table0 count1 table1 ... countN tableN). The initial waveform is given by table0, which is interpolated linearly to table1 over the first count1 samples. From count1 to count2 samples, the waveform is interpolated from table1 to table2, and so on. If more than countN samples are generated, tableN is used for the remainder of the sound. The duration and logical stop time of the sound is taken from fm, which specified frequency modulation (deviation) in Hertz. You should use siosc instead (see Section 7.2.2.1).

7.6.5 Phase Vocoder Functions

snd-phasevocoder(s, map, fftsize, hopsize mode) [SAL]
ncode{(snd-phasevocoder s map fftsize hopsize mode) [LISP]}

Phase vocoder: Identical to phasevocoder except that fftsize, hopsize and mode are not optional. Specify -1 to get the default values for fftsize and hopsize. Specify 0 for the default value of mode. You should use phasevocoder instead (see Section 7.2.2.6).
7.6.6 Physical Model Functions

These functions perform some sort of physically-based modeling synthesis.

```lisp
(snd-bandedwg freq bowpress-env preset sr) [SAL]
(snd-bandedwg freq bowpress-env preset sr) [LISP]
```

A Banded Wave Guide Percussion instrument implemented in STK. The parameter `freq` is a `FLONUM` in Hz, `bowpress-env` is a `SOUND` that ranges from zero to one, `preset` is a `FIXNUM`, and `sr` is the desired sample rate in Hz. Currently, there are four presets: uniform-bar (0), tuned-bar (1), glass-harmonica (2), and tibetan-bowl (3). You should use `wg-uniform-bar`, `wg-tuned-bar`, `wg-glass-harm`, or `wg-tibetan-bowl` instead (see Section 7.2.2.5).

```lisp
(snd-bowed freq bowpress-env sr) [SAL]
(snd-bowed freq bowpress-env sr) [LISP]
```

A bowed string instrument implemented in STK. The `freq` is a `FLONUM` in Hertz, `bowpress-env` is a `SOUND` that ranges from zero to one, and `sr` is the desired sample rate (a `FLONUM`). You should use `bowed` instead (see Section 7.2.2.5).

```lisp
(snd-bowed-freq freq bowpress-env freq-env sr) [SAL]
(snd-bowed-freq freq bowpress-env freq-env sr) [LISP]
```

A bowed model just like `snd-bowed` but with an additional parameter for continuous frequency control. You should use `bowed-freq` instead (see Section 7.2.2.5).

```lisp
(snd-clarinet freq breath-env sr) [SAL]
(snd-clarinet freq breath-env sr) [LISP]
```

A clarinet model implemented in STK. The `freq` is a `FLONUM` in Hertz, `breath-env` is a `SOUND` that ranges from zero to one, and `sr` is the desired sample rate (a `FLONUM`). You should use `clarinet` instead (see Section 7.2.2.5).

```lisp
(snd-clarinet-freq freq breath-env freq-env sr) [SAL]
(snd-clarinet-freq freq breath-env freq-env sr) [LISP]
```

A clarinet model just like `snd-clarinet` but with an additional parameter for vibrato generation and continuous control of reed stiffness and breath noise. You should use `clarinet-freq` instead (see Section 7.2.2.5).

```lisp
(snd-clarinet-all freq vibrato-freq vibrato-gain freq-env breath-env reed-stiffness noise sr) [SAL]
(snd-clarinet-all freq vibrato-freq vibrato-gain freq-env breath-env reed-stiffness noise sr) [LISP]
```

A clarinet model just like `snd-clarinet-freq` but with additional parameters for vibrato generation and continuous control of reed stiffness and breath noise. You should use `clarinet-all` instead (see Section 7.2.2.5).

```lisp
(snd-flute freq breath-env sr) [SAL]
(snd-flute freq breath-env sr) [LISP]
```

A flute implemented in STK. The `freq` is a `FLONUM` in Hertz, `breath-env` is a `SOUND` that ranges from zero to one, and `sr` is the desired sample rate (a `FLONUM`). You should use `flute` instead (see Section 7.2.2.5).
A flute model just like snd-flute but with an additional parameter for continuous frequency control. You should use flute-freq instead (see Section 7.2.2.5).

A flute model just like snd-flute-freq but with additional parameters for vibrato generation and continuous control of breath noise. You should use flute-all instead (see Section 7.2.2.5).

A plucked double-string instrument model implemented in STK. The t0 parameter is the starting time (in seconds), freq is a FLONUM in Hz, body-size and detune are FLONUMs, and sr is the desired sample rate. You should use mandolin instead (see Section 7.2.2.5).

Struck bar instrument model implemented in STK. The parameter t0 is the starting time (in seconds), freq is a FLONUM in Hz, preset is a FIXNUM ranging from 0 to 8, dur is a FLONUM that sets the duration (in seconds) and sr is the desired sample rate. You should use modalbar instead (see Section 7.2.2.5).

A sax model implemented in STK. The freq is a FLONUM in Hertz, breath-env is a SOUND that ranges from zero to one, and sr is the desired sample rate (a FLONUM). You should use sax instead (see Section 7.2.2.5).

A sax model just like snd-sax but with an additional parameter for continuous frequency control. You should use sax-freq instead (see Section 7.2.2.5).

A sax model just like snd-sax-freq but with additional parameters for vibrato generation and continuous control of reed stiffness, breath noise, excitation position, and reed table offset. You should use sax-all instead (see Section 7.2.2.5).
snd-sitar(t0, freq, dur, sr) [SAL]
(snd-sitar t0 freq dur sr) [LISP]
A sitar model implemented in STK. The parameter t0 is the starting time, freq is a FLONUM (in Hz), dur sets the duration and sr is the sample rate (in Hz) of the resulting sound. You should use sitar instead (see Section 7.2.2.5).

7.6.7 Sequence Support Functions

The next two functions are used to implement Nyquist’s seq construct.

snd-seq(sound, closure) [SAL]
(snd-seq sound closure) [LISP]
This function returns sound until the logical stop time of sound. Then, the XLISP closure is evaluated, passing it the logical stop time of sound as a parameter. The closure must return a sound, which is then added to sound. (An add is used so that sound can continue past its logical stop if desired.) Do not call this function. See seq in Section 7.4.

snd-multiseq(array, closure) [SAL]
(snd-multiseq array closure) [LISP]
This function is similar to snd-seq except the first parameter is a multichannel sound rather than a single sound. A multichannel sound is simply an XLISP array of sounds. An array of sounds is returned which is the sum of array and another array of sounds returned by closure. The closure is passed the logical stop time of the multichannel sound, which is the maximum logical stop time of any element of array. Do not call this function. See seq in Section 7.4.
There are many global variables in Nyquist. A convention in Lisp is to place asterisks (*) around global
variables, e.g. *table*. This is only a convention, and the asterisks are just like any other letter as far as
variable names are concerned. Here are some globals users should know about:

*~*=tolerance*
The tolerance used by the SAL "approximately equal" operator (~=) to determine if two numbers are
approximately equal.

*table*
Default table used by osc and other oscillators. Initially set to the sinusoid in *sine-table*.

*A4-Hertz*
Frequency of A4 in Hertz. Note: you must call (set-pitch-names) to recompute pitches after
changing *A4-Hertz*.

*autonorm*
The normalization factor to be applied to the next sound when *autonorm-type* is 'previous. See
Sections 5.3 and 7.5.

*autonormflag*
Enables the automatic normalization feature of the play command. You should use (autonorm-on)
and (autonorm-off) rather than setting *autonormflag* directly. See Sections 5.3 and 7.5.

*autonorm-max-samples*
Specifies how many samples will be computed searching for a peak value when *autonorm-type* is
'lookahead. See Sections 5.3 and 7.5.

*autonorm-previous-peak*
The peak of the previous sound generated by play. This is used to compute the scale factor for the
next sound when *autonorm-type* is 'previous. See Sections 5.3 and 7.5.

*autonorm-target*
The target peak amplitude for the autonorm feature. The default value is 0.9. See Sections 5.3 and 7.5.
*autonorm-type*
   Determines how the autonorm feature is implemented. Valid values are ’lookahead (the default) and ’previous. See Sections 5.3 and 7.5.

*breakenable*
   Controls whether XLISP enters a break loop when an error is encountered. See Section C.14.

*clipping-error*
   If the peak absolute amplitude value of a sound saved or played exceeds *clipping-threshold*, an XLISP error is raised. See *clipping-threshold* for more detail.

*clipping-threshold*
   See *clipping-error* for a description of this variable. *clipping-threshold* is initialized to 127/128. This number is conservative, and it is possible to slightly exceed this value, even with 8-bit files without actual clipping (consider rounding. Also, floating point format files will not clip even when the amplitude exceeds 1.0. Note that a "clipping" threshold of 1.0 is optimistic: 1.0 corresponds to a 16-bit integer value of 32,768 (2^15), but the maximum positive 16-bit integer is 32,767. Thus, a positive sample of 1.0 will clip when written or played as 16-bit audio.

*control-srate*
   Part of the environment, establishes the control sample rate. See Section 3.1 for details.

*default-plot-file*
   The default file for plot data (written by Nyquist, read by NyquistIDE). Default value is "points.dat".

*default-sf-bits*
   The default bits-per-sample for sound files. Typically 16.

*default-sf-dir*
   The default sound file directory. Unless you give a full path for a file, audio files are assumed to be in this directory. (Applies to many functions that deal with sound files. Check the function description to see if *default-sf-dir* applies.)

*default-sf-format*
   The default sound file format. When you write a file, this will be the default format: AIFF for Mac and most Unix systems, NeXT for NeXT systems, and WAV for Win32.

*default-sf-srate*
   The default sample rate for sound files. Typically 44100.0, but often set to 22050.0 for speed in non-critical tasks.

*default-control-srate*
   Default value for *control-srate*. This value is restored when you execute (top) to pop out of a debugging session. Change it by calling (set-control-srate value).
*default-sound-srate*
Default value for *sound-srate*. This value is restored when you execute (top) to pop out of a debugging session. Change it by calling (set-sound-srate value).

*file-separator*
The character that separates directories in a path, e.g. "/" for Unix, ":" for Mac, and "\" for Win32. This is normally set in system.lsp.

*lpslider-cutoff*
The cutoff frequency used by lpslider, described in Section 9.1.3. The default value is 20Hz. If unbound, this variable is set when you load sliders.lsp.

*rslt*
When a function returns more than one value, *rslt* is set to a list of the "extra" values. This provides a make-shift version of the multiple-value-return facility in Common Lisp.

*saw-table*
A single cycle sawtooth wave intended for use by table-lookup oscillators such as osc. This is the waveform used by osc-saw. Note that some aliasing will normally occur when using this waveform, which is not band-limited.

*sine-table*
A single cycle sinusoid intended for use by table-lookup oscillators such as osc.

*snd-device*
Select the audio output device by setting this to a FIXNUM to select a device by index number or to a STRING to select a device by name. If a STRING is provided, the first device whose name contains the STRING (as a substring) is selected. (See *snd-list-devices* below.) If the value is unbound or NULL, the default PortAudio device is used.

*snd-list-devices*
List all audio output devices (as text output) when a sound is played. By inspecting this list, one can determine possible values for *snd-device* and see what device is being selected by Nyquist. One cannot (currently) obtain the list of devices as an XLISP value. The default behavior is to list the devices and some instructions only the first time any sound is played. Setting *snd-list-devices* to a value (t or nil) before playing a sound will override this behavior.

*sound-srate*
Part of the environment, establishes the audio sample rate. See Section 3.1 for details.

*soundenable*
Controls whether writes to a sound file will also be played as audio. Set this variable by calling (sound-on) or (sound-off).

*tracenable*
Controls whether XLISP prints a backtrace when an error is encountered.
*tri-table*

A single cycle triangle wave intended for use by table-lookup oscillators such as osc. This is the waveform used by osc-tri. Note that some aliasing will normally occur when using this waveform, which is not band-limited.

**XLISP variables**

See Section C.14 for a list of global variables defined by XLISP.

**Environment variables**

See Section 3.1 for definitions of variables used in the environment for behaviors. In general, you should never set or access these variables directly.

**Various constants**

See Section 1.8 for definitions of predefined constants for loudness, duration, and pitch.
9 Interactive Nyquist

Nyquist is not intended for real-time performance, but it has some features that allow you to adjust parameters interactively. The basic idea is that there is an array of 1000 floating point values, called sliders, that can be accessed while synthesizing sounds in Nyquist. The \texttt{slider} unit generator returns a signal that copies the current value of a slider value. You can change the slider value while playing sounds using either Open Sound Control or the NyquistIDE.

Sounds are normally computed on demand. So the result returned by \texttt{slider} does not immediately compute any samples. Samples are only computed when something tries to use this signal. At that time, the slider value is read. Normally, if the slider is used to control a sound, you will hear changes in the sound pretty soon after the slider value changes. However, one thing that can interfere with this is that \texttt{SOUND} samples are computed in blocks of about 1000 samples. When the slider value is read, the same value is used to fill a block of 1000 samples, so even if the sample rate is 44,100 Hz, the effective slider sample rate is 44,100/1000, or 44.1 Hz. If you give the slider a very low sample rate, say 1000, then slider value changes will only be noticed by Nyquist approximately once per second. For this reason, you should normally use the audio sample rate (typically 44,100 Hz) for the rate of the \texttt{snd-slider} output \texttt{SOUND}. (Yes, this is terribly wasteful to represent each slider value with 1000 samples, but Nyquist was not designed for low-latency computation, and this is an expedient work-around.)

When you load \texttt{sliders.lsp}, which defines a number of slider functions, two important settings may be changed. First \texttt{autonorm-off} is called. The problem with auto-normalization is that it works by computing 1 million samples ahead of real time to determine a normalization factor. If Nyquist computes ahead, it will be unable to respond to control changes until the million samples (about 20 seconds) have been played. Secondly, \texttt{snd-set-latency} is used to set the audio latency to 0.02s (20 milliseconds). Normally, Nyquist uses a generous 0.3s latency which allows Nyquist to stop computing audio and run garbage collection without breaks in the audio output. At 20ms, interactivity is greatly enhanced because changes do not sit in audio buffers for 300ms, but you may notice some break-up in the audio, especially when garbage collection takes place. The latency can be changed to any value you like after you load \texttt{sliders.lsp}.

In addition to reading sliders as continually changing \texttt{SOUND}s, you can get the slider value as a Lisp \texttt{FLONUM} (a floating point number) using \texttt{get-slider-value}. This might be useful if you are computing a sequence of many notes (or other sound events) and want to apply the current slider value to the whole note or sound event.

Other unit generators exist to instantiate behaviors and to stop sounds according to slider values. These will be described below.
9.1 Interactive Control with the NyquistIDE

To control sounds interactively using the NyquistIDE, you first create a control panel, then populate the panel with sliders and buttons. These will send values into the sliders array in the Nyquist process. To control a sound, you use built-in functions to retrieve real-time values from the sliders array as sound is being played. Further discussion and examples can be found in demos/slider-demos.sal.

9.1.1 Creating a Control Panel

A control panel is created with make-slider-panel, which takes a panel name and color as parameters. Control panels can only be created by executing code in Nyquist. There is no way to configure control panels directly using the NyquistIDE. Control panels can be deleted interactively using the close button on the panel or through code by calling close-slider-panel.

```
(make-slider-panel name color)
```

Create a control panel in the IDE. The title of the control panel window is given by name, a STRING. The color of the panel is given by color, a FIXNUM from 0 through 12. The color 0 is gray. Other colors are implementation-dependent, but different numbers give distinguishable colors. You must load sliders.lsp to access this function.

```
(close-slider-panel name)
```

Close a control panel in the NyquistIDE named by name, a STRING. Any embedded controls (sliders or buttons) are also destroyed. You must load sliders.lsp to access this function.

9.1.2 Creating Controls

You can create slider and button controls. A slider control adjusts a floating point value in the sliders array and accessible as a time-varying signal (a SOUND) or as a FLONUM. A button control sets a floating point value in the sliders array to zero (0.0) or one (1.0). The value changes when the left mouse button is pressed over the button control and changes back when the mouse button is released.

```
(make-slider name init low high)
```

Create a slider in the most recently created control panel. (Thus, you should populate a control panel with sliders and buttons before creating another control panel.) Sliders have a label specified by name, a STRING, an initial value specified by init, a FLONUM, a minimum value specified by low, a FLONUM, and a maximum value specified by high, a FLONUM. Sliders are added to the current panel in order from top to bottom. You must load sliders.lsp to access this function.
Nyquist Reference Manual

Chapter 9. Interactive Nyquist

make-button(name [, normal]) [SAL]
(make-button name [normal]) [LISP]

Create a button in the most recently created control panel. Buttons have a label specified by name, a STRING, and a "normal value" specified by 0 or 1 (a FIXNUM). If 0 is specified, the value controlled by the button is 0 when the button is released and 1 when the button is pressed. If 1 is specified, the normal value is 1 changing to 0 when the button is pressed. Buttons are added to the control panel in order from top to bottom. You must load sliders.lsp to access this function.

9.1.3 Accessing Control Values

Each control created by make-slider or make-button is assigned a slider index from 10 to 999. Control changes are passed via hidden text input from the NyquistIDE to the nyquist process, where the values are converted to floats and stored in the slider array. You can then access these values with either slider, lpslider, or get-slider-value.

slider(number [, dur]) [SAL]
slider(name [, dur]) [SAL]
slider(panel, name [, dur]) [SAL]
(slider number [dur]) [LISP]
(slider name [dur]) [LISP]
(slider panel name [dur]) [LISP]

Create a SOUND that reads signal values from the slider array. In the first form, the first parameter is the index (a FIXNUM) of the value in the slider array. In the second form, the slider value will be controlled by the NyquistIDE control created by make-slider or make-button using the same name, a STRING. The control must be in the most recently created panel. In the third form, the panel named panel is searched for the control named name to determine the value. In all cases, the optional dur, a FLONUM, is used to determine the duration of the sound. This duration is scaled by the environment in the usual way. You must load sliders.lsp to access this function.

lpslider(number [, dur]) [SAL]
lpslider(name [, dur]) [SAL]
lpslider(panel, name [, dur]) [SAL]
lpslider number [dur]) [LISP]
lpslider name [dur]) [LISP]
lpslider panel name [dur]) [LISP]

Create a SOUND based on the value of an interactive control. This function is exactly like slider, except the sound is low-pass filtered to avoid sudden jumps when the control value is adjusted. The low-pass filter cutoff is determined by *lpslider-cutoff*, which is initialized to 20Hz when slider.lsp is loaded. You must load sliders.lsp to access this function.

get-slider-value(number) [SAL]
get-slider-value(name) [SAL]
get-slider-value(panel, name) [SAL]
(get-slider-value number) [LISP]
(get-slider-value name) [LISP]
(get-slider-value panel name) [LISP]

Get a value (a FLONUM) stored in the slider array. The array is accessed directly if the parameter is number, a FIXNUM. The index can be determined by searching the most recently created control panel by name, a
STRING, if only name is given. If both panel and name are given (STRINGS), the named panel is searched for the named control. The result is a FLONUM. You must load sliders.lsp to access this function.

```
snd-slider(index, t0, srate, duration) [SAL]
(snd-slider index t0 srate duration) [LISP]
```

Create a sound controlled by the slider named by index (an integer index into the array of sliders). The function returns a sound. Since Nyquist sounds are computed in blocks of samples, and each block is computed at once, each block will contain copies of the current slider value. Normally, you would call slider (see above) rather than this low-level function.

### 9.1.4 Starting and Stopping Sounds

All Nyquist sounds have a duration. Even the slider unit generator has a duration and terminates at the end of that duration. In most cases, you will instead want interactive functions to run until you interactively ask them to stop. The stop-on-zero function terminates when an input signal (typically a slider) goes to zero. This can be used to terminate a complex interactively controlled sound. For example, here is a tone that terminates when the Stop button is pressed:

```
exec make-button("Stop", 1)

function can-stop()
    play (hzosc(1000) * stop-on-zero(slider("Stop"))) ~ 100
```

The stretch factor of 100 will cause this tone to play for 100 seconds. However, if the button named "Stop" is pressed, it will change value from the "normal" value 1 to 0. When the signal goes to zero, stop-on-zero will terminate. The multiplication then immediately terminates (because anything multiplied by zero will be zero; termination is just an efficient way to return zeros from now on). Since the multiplication is the top-level sound being played, the play stops.

Another thing you might want to do with interactive control is start some sound. The trigger function computes an instance of a behavior each time an input SOUND goes from zero to greater-than-zero. This can be used, for example, to create a sequence of sound events interactively. For example:

```
exec make-button("Trigger", 0)

function trigger-me()
    play trigger(slider("Trigger", 100), pluck(c3))
```

Here, a button control changes value from 0 to 1 when the button is pressed. This value is retrieved by the slider function, which runs for 100 seconds. When the button and hence the slider goes from 0 to 1, the behavior, pluck(c3) is instantiated. Many instances can be triggered by the button.

```
stop-on-zero(s) [SAL]
(stop-on-zero s) [LISP]
```

Return a SOUND that is identical to s, a SOUND, except the returned sound terminates when s first goes to zero. When a sound terminates, it remains at zero. A SOUND multiplication terminates when either parameter terminates, so multiplying by stop-on-zero is a way to terminate a sound interactively. (See the example above.) You must load sliders.lsp to access this function.
trigger(s, beh) [SAL]
(trigger s beh) [LISP]
Returns a sound which is the sum of zero or more possibly overlapping instances of the behavior beh. One instance is created each time SOUND s makes a transition from less than or equal to zero to greater than zero. (If the first sample of s is greater than zero, an instance is created immediately.) The sample rate of s and all behaviors must be the same, and the behaviors must be (monophonic) SOUNDs. The sample rate of the result is also the sample rate of s. The stop time of the result is the maximum stop time of s and all sounds returned by instances of the behavior. This function is particularly designed to allow behaviors to be invoked in real time. See the trigger-me function definition shown above.

An implementation note: There is no way to have trigger return a multichannel sound. An alternative implementation would be a built-in function to scan ahead in a sound to find the time of the next zero crossing. This could be combined with some LISP code similar to seq to sum up instances of the closure. However, this would force arbitrary look-ahead and therefore would not work with real-time inputs, which was the motivation for trigger in the first place.

snd-stoponzero(s) [SAL]
(snd-stoponzero s) [LISP]
This function is identical to stop-on-zero. You should use stop-on-zero instead.

snd-trigger(s, closure) [SAL]
(snd-trigger s closure) [LISP]
This is a low-level support function for trigger. The closure takes a starting time and returns a SOUND. See trigger above for more details. Use trigger as described above and do not call this function directly.

### 9.2 Using Open Sound Control

Open Sound Control (OSC) is a simple protocol for communicating music control parameters between software applications and across networks. For more information, see [http://www.cnmat.berkeley.edu/OpenSoundControl/](http://www.cnmat.berkeley.edu/OpenSoundControl/).

The Nyquist implementation of Open Sound Control is simple: an array of floats can be set by OSC messages and read by Nyquist functions. That is about all there is to it.

The slider and get-slider-value functions, described above, can be used to access these values within Nyquist. Each of these functions can take a slider array index to specify which value to use. Since make-slider allocates slider indices starting at 10, it is recommended that you control sliders 0 through 9 via OSC. If you change a slider array value via OSC that is already controlled by a graphical slider in the NyquistIDE, the graphical slider will not be updated or synchronized to the OSC value. (And there is no current way to send OSC command to the NyquistIDE.)

Note: Open Sound Control must be enabled by calling osc-enable(t). If this fails under Windows, see the installation instructions in `sys/win/README.txt` regarding SystemRoot.

osc-enable(flag) [SAL]
(osc-enable flag) [LISP] Enable or disable Open Sound Control. (See Section 9.2.) Enabling creates a socket and a service that listens for UDP packets on port 7770. Currently, only two messages are accepted by Nyquist. The first is of the form /slider with an integer index and a floating point value. These set internal slider values accessed by the slider and get-slider-value functions. The second is of the form /wii/orientation with two floating point values. This message is a special case to support the DarwinRemoteOsc program which can relay data from a Nintendo WiiMote device to Nyquist via OSC. The two orientation values control sliders 0 and 1. Disabling terminates the service (polling for messages) and closes the socket. The previous state of enablement is returned, e.g. if OSC is
enabled and flag is nil, OSC is disabled and T (true) is returned because OSC was enabled at the time of the call. This function only exists if Nyquist is compiled with the compiler flag OSC. Otherwise, the function exists but always returns the symbol DISABLED. Warning: there is the potential for network-based attacks using OSC. It is tempting to add the ability to evaluate XLISP expressions sent via OSC, but this would create unlimited and unprotected access to OSC clients. For now, it is unlikely that an attacker could do more than manipulate slider values.

9.2.1 Sending Open Sound Control Messages

A variety of programs support OSC. The only OSC message interpreted by Nyquist has an address of /slider, and two parameters: an integer slider number and a float value, nominally from 0.0 to 1.0.

Two small programs are included in the Nyquist distribution for sending OSC messages. (Both can be found in the same directory as the nyquist executable.) The first one, osc-test-client sends a sequence of messages that just cause slider 0 to ramp slowly up and down. If you run this on a command line, you can use "?" or "h" to get help information. There is an interactive mode that lets you send each OSC message by typing RETURN.

9.2.2 The ser-to-osc Program

The second program is ser-to-osc, a program that reads serial input (for example from a PIC-based microcontroller) and sends OSC messages. Run this command-line program from a shell (a terminal window under OS X or Linux; use the CMD program under Windows). You must name the serial input device on the command line, e.g. under OS X, you might run:

```bash
./ser-to-osc /dev/tty.usbserial-0000103D
```

(Note that the program name is preceded by "./". This tells the shell exactly where to find the executable program in case the current directory is not on the search path for executable programs.) Under Windows, you might run:

```
ser-to-osc com4
```

(Note that you do not type "./" in front of a windows program.)

To use ser-to-osc, you will have to find the serial device. On the Macintosh and Linux, try the following:

```
ls /dev/*usb*
```

This will list all serial devices with "usb" in their names. Probably, one will be a name similar to /dev/tty.usbserial-0000103D. The ser-to-osc program will echo data that it receives, so you should know if things are working correctly.

Under Windows, open Control Panel from the Start menu, and open the System control panel. Select the Hardware tab and click the Device Manager button. Look in the device list under Ports (COM & LPT). When you plug in your serial or USB device, you should see a new entry appear, e.g. COM4. This is the device name you need.

The format for the serial input is: any non-whitespace character(s), a slider number, a slider value, and a newline (control-j or ASCII 0x0A). These fields need to be separated by tabs or spaces. An optional carriage return (control-m or ASCII 0x0D) preceding the ASCII 0x0A is ignored. The slider number should be in decimal, and the slider value is a decimal number from 0 to 255. This is scaled to the range 0.0 to 1.0 (so an input of 255 translates to 1.0).

There is a simple test program in demos/osc-test.lsp you can run to try out control with Open Sound Control. There are two examples in that file. One uses snd-slider to control the frequency of an oscillator. The other uses get-slider-value to control the pitch of grains in a granular synthesis process.
10 Time/Frequency Transformation

Nyquist provides functions for FFT and inverse FFT operations on streams of audio data. Because sounds can be of any length, but an FFT operates on a fixed amount of data, FFT processing is typically done in short blocks or windows that move through the audio. Thus, a stream of samples is converted into a sequence of FFT frames representing short-term spectra.

Nyquist does not have a special data type corresponding to a sequence of FFT frames. This would be nice, but it would require creating a large set of operations suitable for processing frame sequences. Another approach, and perhaps the most "pure" would be to convert a single sound into a multichannel sound, with one channel per bin of the FFT.

Instead, Nyquist violates its "pure" functional model and resorts to objects for FFT processing. A sequence of frames is represented by an XLISP object. Whenever you send the selector :next to the object, you get back either NIL, indicating the end of the sequence, or you get an array of FFT coefficients.

The Nyquist function snd-fft (mnemonic, isn’t it?) returns one of the frame sequence generating objects. You can pass any frame sequence generating object to another function, snd-ifft, and turn the sequence back into audio.

With snd-fft and snd-ifft, you can create all sorts of interesting processes. The main idea is to create intermediate objects that both accept and generate sequences of frames. These objects can operate on the frames to implement the desired spectral-domain processes. Examples of this can be found in the file demos/fft_tutorial.htm, which is part of the standard Nyquist release. The documentation for snd-fft and snd-ifft follows.

\[
\texttt{snd-fft}(\texttt{sound}, \texttt{length}, \texttt{skip}, \texttt{window}) \quad \text{[SAL]} \\
(\texttt{snd-fft sound length skip window}) \quad \text{[LISP]}
\]

This function performs an FFT on the first samples in \texttt{sound} and returns a Lisp array of \texttt{FLONUM}s. The function modifies the \texttt{sound}, violating the normal rule that sounds are immutable in Nyquist, so it is advised that you copy the sound using \texttt{snd-copy} if there are any other references to \texttt{sound}. The length of the FFT is specified by \texttt{length}, a \texttt{FIXNUM} (integer) which must be a power of 2. After each FFT, the sound is advanced by \texttt{skip} samples, also of type \texttt{FIXNUM}. Overlapping FFTs, where \texttt{skip} is less than \texttt{length}, are allowed. If \texttt{window} is not NIL, it must be a sound. The first \texttt{length} samples of \texttt{window} are multiplied by \texttt{length} samples of \texttt{sound} before performing the FFT. When there are no more samples in \texttt{sound} to transform, this function returns NIL. The coefficients in the returned array, in order, are the DC coefficient, the first real, the first imaginary, the second real, the second imaginary, etc. The last array element corresponds to the real coefficient at the Nyquist frequency.

\[
\texttt{snd-ifft}(\texttt{time}, \texttt{srate}, \texttt{iterator}, \texttt{skip}, \texttt{window}) \quad \text{[SAL]} \\
(\texttt{snd-ifft time srate iterator skip window}) \quad \text{[LISP]}
\]

This function performs an IFFT on a sequence of spectral frames obtained from \texttt{iterator} and returns a sound. The start time of the sound is given by \texttt{time}. Typically, this would be computed by calling \texttt{(local-to-global 0)}. The sample rate is given by \texttt{srate}. Typically, this would be \texttt{*sound-srate*}, but it might also depend upon the sample rate of the sound from which the spectral frames were derived. To obtain each frame, the function sends
the message :next to the iterator object, using XLISP’s primitives for objects and message passing. The object should return an array in the same format as obtained from snd-fft, and the object should return NIL when the end of the sound is reached. After each frame is inverse transformed into the time domain, it is added to the resulting sound. Each successive frame is added with a sample offset specified by skip relative to the previous frame. This must be an integer greater than zero and less than the frame (FFT) size. If window is not NIL, it must be a sound. This window signal is multiplied by the inverse transformed frame before the frame is added to the output sound. The length of each frame should be the same power of 2. The length is implied by the first array returned by iterator, so it does not appear as a parameter. This length is also the number of samples used from window. Extra samples are ignored, and window is padded with zeros if necessary, so be sure window is the right length. The resulting sound is computed on demand as with other Nyquist sounds, so :next messages are sent to iterator only when new frames are needed. One should be careful not to reuse or modify iterator once it is passed to snd-iff. 
11 MIDI, Adagio, and Sequences

Nyquist includes facilities to read and write MIDI files as well as an ASCII text-based score representation language, Adagio. XLISP and Nyquist can be used to generate MIDI files using compositional algorithms. (See also Section 14.) A tutorial on using the Adagio representation and MIDI can be found in demos/midi_tutorial.htm. The Adagio language is described below. Adagio was originally developed as part of the CMU MIDI Toolkit, which included a program to record and play MIDI using the Adagio representation. Some of the MIDI features of Adagio may not be useful within Nyquist.

Nyquist offers a number of different score representations, and you may find this confusing. In general, MIDI files are a common way to exchange music performance data, especially with sequencers and score notation systems. The demos/midi_tutorial.htm examples show how to get the most precise control when generating MIDI data. Adagio is most useful as a text-based score entry language, and it is certainly more compact than Lisp expressions for MIDI-like data. The Xmusic library (Chapter 14) is best for algorithmic generation of music and score manipulation. There are functions to convert between the Adagio, MIDI sequence data, and Xmusic score representations.

11.1 The SEQ Type

Nyquist has a special data type to store MIDI data called a SEQ (short for "sequence"). You can create an empty SEQ object, read data into it from a MIDI or Adagio file, write a SEQ as MIDI or Adagio, and insert notes into a SEQ object. You can also convert SEQ objects into sound by providing functions to handle different MIDI messages. Further discussion and examples can be found in demos/midi_tutorial.htm.

seq-create() [SAL]
(seq-create) [LISP]
    Creates a SEQ data object.

seq-read(seq, file) [SAL]
(seq-read seq file) [LISP]
    Reads into a SEQ data object from an ASCII text file in Adagio format. The first parameter is the SEQ object to read. The file is a file opened for reading. It must be closed after seq-read returns.

seq-read-smf(seq, midi-file) [SAL]
(seq-read-smf seq midi-file) [LISP]
    Reads into a SEQ data object from a Standard MIDI File. The first parameter is the SEQ object to read. The midi-file is a binary file opened in binary mode for reading. It must be closed after seq-read-smf returns.
### 11.2 Adagio Score Language

Adagio is an easy-to-use, non-procedural notation for scores. In Adagio, text commands are used to specify each note. If you are new to Adagio, you may want to glance at the examples in Section 11.5 starting on page 149 before reading any further.

A note is described in Adagio by a set of attributes, and any attribute not specified is "inherited" from the previous line. Attributes may appear in any order and must be separated by one or more blanks. An attribute may not contain any blanks. The attributes are: time, pitch, loudness, voice number, duration, and articulation.

Adagio has been used to program a variety of hardware and software synthesizers, and the Adagio compiler can be easily adapted to new environments. Although not originally intended for MIDI, Adagio works quite well as a representation for MIDI scores. Adagio has been extended to allow MIDI controller data such as modulation wheels, pitch bend, and volume, MIDI program commands to change timbre, and System Exclusive messages.

A note command in Adagio must be separated from other notes. Usually, notes are distinguished by writing each one on a separate line. Notes can also be separated by using a comma or semicolon as will be described below.

Besides notes, there are several other types of commands:
1. An asterisk (*) in column one (or immediately after a comma, semicolon, or space) indicates that the rest of the line is a comment. The line is ignored by Adagio, and is therefore a good way to insert text to be read by people. Here are some examples:
   * This is a comment.
   T150 G4  * This is a comment too!
   T150 G4  ;* So is this.

2. An empty command (a blank line, for example) is ignored as if it were a comment\(^1\).

3. An exclamation point (!) in column one (or immediately after a comma or semicolon) indicates a special command. A special command does not generate a note. Special commands follow the "!" with no intervening spaces and extend to the end of the line, for example:
   !TEMPO 100

4. Control change commands are used to control parameters like pitch bend, modulation, and program (timbre). Control change commands can be specified along with notes or by themselves. A command that specifies control changes without specifying a pitch will not produce a note.

Adagio is insensitive to case, thus "A" is equivalent to "a", and you can mix upper and lower case letters freely.

### 11.3 Specifying Attributes

A note is indicated by a set of attributes. Attributes are indicated by a string of characters with no intervening spaces because spaces separate attributes. The attributes are described below.

The default unit of time is a centisecond (100\(^{th}\)'s), but this can be changed to a millisecond (1000\(^{th}\)'s) using the !MSEC command and reset to centiseconds with !CSEC (see Section 11.6.1). In the descriptions below, the term "time unit" will be used to mean whichever convention is currently in effect.

#### 11.3.1 Time

The time attribute specifies when to start the note. A time is specified by a "T" followed by a number representing time units or by a duration (durations are described below). Examples:

T150 ** 1.5 sec (or .15 sec)
TQ3 ** 3 quarter note's duration

If no time is specified, the default time is the sum of the time and duration attributes of the previous note. (But see Section 11.3.4.) Time is measured relative to the time of the most recent Tempo or Rate command. (See the examples in Section 11.5 for some clarification of this point.)

\(^1\)To be consistent, a blank line ought to specify zero attributes and generate a note that inherits all of its attributes from the previous one. Adagio is intentionally inconsistent in this respect.
11.3.2 Pitch

The pitch attribute specifies what frequency to produce. Standard scale pitches are named by name, using S for sharp, F for flat, and (optionally) N for natural. For example, C and CN represent the same pitch, as do FS and GF (F sharp and G flat). Note that there are no bar lines, and accidentals do not carry forward to any other notes as in common practice notation.

Octaves are specified by number. C4 is middle C, and B3 is a half step lower. F5 is the top line of the treble clef, etc. (Adagio octave numbering follows the ISO standard, but note that this is not universal. In particular, Yamaha refers to middle C as C3.) Accidentals can go before or after the octave number, so FS3 and F3S have the same meaning.

An alternate notation for pitch is Pn, where n is an integer representing the pitch. Middle C (C4) is equivalent to P60, CS4 is P61, etc.

If you do not specify an octave, Adagio will choose one for you. This is done by picking the octave that will make the current pitch as close to the previous pitch as possible. In the case of augmented fourths or diminished fifths, there are two equally good choices. Adagio chooses the lower octave.

11.3.3 Duration

Duration is specified by a letter indicating a number of beats, followed by one or several modifiers. The basic duration codes are:

- W (whole, 4 beats),
- H (half, 2 beats),
- Q (quarter, 1 beat),
- I (eighth, 1/2 beat),
- S (sixteenth, 1/4 beat),
- % (thirtysecond, 1/8 beat),
- ^ (sixtyfourth, 1/16 beat).

Note that E is a pitch, so eighth-notes use the duration code I. The default tempo is 100 beats per minute (see Section 11.3.10). These codes may be followed by a T (triplet), indicating a duration of 2/3 the normal. A dot (.) after a duration code extends it by half to 3/2 the normal. An integer after a note multiplies its duration by the indicated value (the result is still just one note). Finally, a slash followed by an integer divides the duration by the integer. Like all attributes, duration attributes may not have embedded spaces. Examples:

- Q 1 beat (quarter note)
- QT 2/3 beat (quarter triplet)
- W 6 beats (dotted whole note)
- ST6 1 beat (6 sixteenth triplets)
- H5 10 beats (5 half notes)
- Q3/7 3/7 beats

A duration may be noted by Un, where n is an integer indicating 100th’s of a second (or 1000th’s), see Section 11.6.1. For example, U25 is twenty-five time units.

Durations may be combined using a plus sign:
11.3.4 Next Time

The time of the next command (the next command in the Adagio program text) is normally the time of the current note command plus the duration of the current note. This can be overridden by a field consisting of the letter N followed by a number indicating time units, or followed by a duration as described above. The next note will then start at the time of the current note plus the duration specified after N. If the next note has an explicit time attribute (T), then the specified time will override the one based on the previous note. Examples:

- N0: start the next note at the same time as this one
- N50: start the next note 0.5 seconds after this one
- NQT: start the next note 2/3 beat after the current one
- NU10+Q: start after 0.1 seconds plus a quarter

A comma has an effect similar to N0 and is explained in Section 11.6.2. Articulation effects such as staccato can be produced using N, but it is more convenient to use the articulation attribute described in Section 11.3.6.

11.3.5 Rest

Rests are obtained by including the field R in a note command. The effect of an R field is to omit the note that would otherwise occur as the result of the current note command. In all other respects, the command is processed just like any other line. This means that attributes such as duration, loudness, and pitch can be specified, and anything specified will be inherited by the note in the next command. Normally, a rest will include just R and a duration. The fact that a note command specifies a rest is not inherited. For example:

- RH: a half (two beat) rest
- RH: illegal, R must be separated from H by space(s)

Because some synthesizers (e.g. a DX7) cannot change programs (presets) rapidly, it may be desirable to change programs in a rest so that the synthesizer will be ready to play by the end of the rest. See Section 11.3.9 for an example.

11.3.6 Articulation

Articulation in Adagio refers to the percentage of time a note is on relative to the indicated duration. For example, to play a note staccato, you would normally play the note about half of its indicated duration. In Adagio, articulation is indicated by # followed by an integer number indicating a percentage. The articulation attribute does not affect the time of the next command. This example plays two staccato quarter notes:

```
C Q #50
D
```
To produce overlapping notes, the articulation may be greater than 100.

Be aware that overlapping notes on the same pitch can be a problem for some synthesizers. The following example illustrates this potential problem:

```
!TEMPO 60
C Q #160 * starts at time 0, ends at 1.6 sec
D I * starts at time 1, ends at 1.8 sec
C Q * starts at time 1.5, ends at 3.1 sec?
```

At one beat per second (tempo 60), these three notes will start at times 0, 1, and 1.5 seconds, respectively. Since these notes have an articulation of 160, each will be on 160% of its nominal duration, so the first note (C) will remain on until 1.6 seconds. But the third note (another C) will start at time 1.5 seconds. Thus, the second C will be started before the first one ends. Depending on the synthesizer, this may cancel the first C or play a second C in unison. In either case, a note-off message will be sent at time 1.6 seconds. If this cancels the second C, its actual duration will be 0.1 rather than 1.6 seconds as intended. A final note-off will be sent at time 3.1 seconds.

### 11.3.7 Loudness

Loudness is indicated by an L followed by a dynamic marking from the following: PPP, PP, P, MP, MF, F, FF, FFF. Alternatively, a number from 1 to 127 may be used. The loudness attribute is the MIDI note velocity. (Note that a MIDI velocity of 0 means "note-off," so the minimum loudness is 1.) The dynamic markings are translated into numbers as follows:

<table>
<thead>
<tr>
<th>Dynamic</th>
<th>Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lppp</td>
<td>20</td>
</tr>
<tr>
<td>Lpp</td>
<td>26</td>
</tr>
<tr>
<td>Lp</td>
<td>34</td>
</tr>
<tr>
<td>Lmp</td>
<td>44</td>
</tr>
<tr>
<td>Lmf</td>
<td>58</td>
</tr>
<tr>
<td>Lf</td>
<td>75</td>
</tr>
<tr>
<td>Lff</td>
<td>98</td>
</tr>
<tr>
<td>Lfff</td>
<td>127</td>
</tr>
</tbody>
</table>

### 11.3.8 Voice

The voice attribute tells which of the 16 MIDI channels to use for the note. The voice attribute consists of a V followed by an integer from 1 (the default) to 16.

There is a limit to how many notes can be played at the same time on a given voice (MIDI channel). Since the limit depends upon the synthesizer, Adagio cannot tell you when you exceed the limit. Similarly, Adagio cannot tell whether your synthesizer is set up to respond to a given channel, so there is no guarantee that what you write will actually be heard.

### 11.3.9 Timbre (MIDI Program)

A MIDI program (synthesizer preset) can be selected using the attribute Zn, where n is the program number (from 1 to 128). Notice that in MIDI, changing the program on a given channel will affect all notes on that channel and possibly others. Adagio treats MIDI program changes as a form of control change.

For many synthesizers, you will not be able to change programs at the start of a note or during a note. Change the program during a rest instead. For example:
Check how your synthesizer interprets program numbers. For example, the cartridge programs on a DX7 can be accessed by adding 32 to the cartridge program number. Cartridge program number 10 is specified by Z42.

As in MIDI, the Adagio timbre is a property of the voice (MIDI channel), so the timbre will not be inherited by notes on a different channel; to change the timbre on multiple voices (channels), you must explicitly notate each change.

### 11.3.10 Tempo

The length of a beat may be changed using a Tempo command:

```
!TEMPO n
```

where \( n \) indicates beats per minute. The exclamation mark tells Adagio that this is a special command line rather than a note definition. A special command takes the place of a note specification. No other attributes should be written on a line with a special command. The `!TEMPO` command is associated with a time, computed as if the `!TEMPO` command were a note. The time attribute (T) of all succeeding notes is now measured relative to the time of the `!TEMPO` command. The new tempo starts at the `!TEMPO` command time and affects all succeeding notes. Durations specified in time units (for example `U58`, `N15`) are not affected by the `!TEMPO` command, and numerical times (for example `T851`) are computed relative to the time of the last `!TEMPO` command.

The `!TEMPO` command is fairly clever about default durations. If the last duration specified before the `!TEMPO` command is symbolic (using one of `\^`, `/`, `s`, `q`, `h`, or `w`), then the default duration for the node after the `!TEMPO` command will be modified according to the tempo change. Consider the following tempo change:

```
!TEMPO 60
A4 H
!TEMPO 120
G
```

In this example, the first note will last 2 seconds (2 beats at 60 beats per minute). The second note inherits the duration (H) from the first note, but at 120 beats per minute, the second note will last only 1 second. If the duration had been specified U200 (also a duration of 2 seconds), the second note would also last 2 seconds because the `!TEMPO` command does not affect times or durations specified numerically in time units. If the duration is the sum of a symbolic and a numeric specification, the inherited duration after a `!TEMPO` command is undefined.

### 11.3.11 Rate

The `!RATE` command scales all times including those specified in hundredths of seconds. A rate of 100 means no change, 200 means twice as fast, and 50 means half as fast. For example, to make a piece play 10% faster, you can add the following command at the beginning of the score:

```
!RATE 110
```

`!RATE` and `!TEMPO` commands combine, so

```
!RATE 200
!TEMPO 70
```

will play 70 beats per minute at double the normal speed, or 140 beats per minute. Like `!TEMPO`, the time of the `!RATE` command is added to the time attribute of all following notes up to the next `!TEMPO` or `!RATE` command.
Two !RATE commands do not combine, so a !RATE command only affects the rate until the next !RATE command.

Although !TEMPO and !RATE can occur in the middle of a note (using N, T, etc.) they do not affect a note already specified. This property allows multiple tempi to exist simultaneously (see Section 11.6.4).

### 11.4 Default Attributes

If an attribute is omitted, the previous one is used by default (with the exception of the time attribute). The default values for the first note, which are inherited by succeeding notes until something else is specified, are given below in Adagio notation:

- **Time**: T0
- **Pitch**: C4
- **Duration**: Q
- **Articulation**: #100
- **Loudness**: LFFF
- **Voice**: V1
- **Tempo**: !TEMPO 100
- **Rate**: !RATE 100

Control changes (including timbre or MIDI program, specified by Z) have no default value and are only sent as specified in the score.

**Important:** the rules for determining when a command will play a note are as follows (and this has changed slightly from previous versions):

1. If a special (!) command or nothing is specified, e.g. a blank line, do *not* play a note.
2. If R (for "rest") is specified, do *not* play a note.
3. Otherwise, if a pitch is specified, *do* play a note.
4. Otherwise, if no control changes (or program changes) are specified (so this is a command with non-pitch attributes and no control changes), *do* play a note.

Another way to say this is "Special commands and commands with rests (R) do not play notes. Otherwise, play a note if a pitch is specified or if no control is specified."

### 11.5 Examples

The following plays the first two bars of "Happy Birthday". Note that Adagio knows nothing of bar lines, so the fact that the first note occurs on beat 3 or that the meter is three-four is of no consequence:
*Example 1 ** Happy Birthday tune (C major)
!TEMPO 120
G4 I. LF
G4 S
A4 Q
G4
C5
B4 H

The time attribute for the first note is zero (0). The second note will occur a dotted eighth later, etc. Notice that no timbre or rate was specified. Adagio will provide reasonable default values of 1 and 100, respectively.

The following example plays the first four bars of an exercise from Bartok’s Mikrokosmos (Vol. 1, No. 12). An extra quarter note is inserted at the beginning of each voice in order to allow time to change MIDI programs. The right hand part is played on voice (MIDI channel) 1 and the left hand part on voice 2. Notice the specification of the time attribute to indicate that voice 2 starts at time 0. Also, default octaves are used to reduce typing.

*Example 2 ** Bartok
*voice 1, right hand
R Q Z10 V1  ** extra rest for program change
A4 H
B Q
C
D H
E
D Q
F
C
G
D
R

*voice 2, left hand
T0 R Q Z15 V2  ** extra rest for program change
G3 H
F Q
E
D H
E
D Q
F
G
F
E
D
R
The next example is the same piece expressed in a different manner, illustrating the interaction between the !TEMPO command and the time attribute. Recall that the time attribute is measured relative to the time of the last !TEMPO command:

```
*Example 3 ** 4 measures in 2 sections
!Tempo 100
*Voice 1, Measures 1 & 2
R Q Z10 V1
A4 H
B Q
C
D H
C

*Voice 2, Measures 1 & 2
T0 R Q Z15 V2
G3 H
F Q
E
D H
E H

!TEMPO 100
*Voice 1, Measures 3 & 4
* note that Z10 is still in effect for V1
V1 D4 Q
C
B
A
B
C
D
R

*Voice 2, Measures 3 & 4
T0 V2 D3 Q
E
F
G
E
D
R
```

The piece is written in 4 sections. The first plays a rest followed by two measures, starting at time 0. The next section changes the time back to zero and plays two measures of the left hand part (voice 2). The next command (!TEMPO 100) sets the tempo to 100 (it already is) and sets the reference time to be two measures into the piece. Therefore, the next note (D4) will begin measure 3. The D3 that begins the last group of notes has a T0 attribute, so it will also start at measure 3. Notice how the !TEMPO command can serve to divide a piece into sections.

The last example will show yet another way to express the same piece of music using the "Next" attribute. Only the first bar of music is given.
*Example 4 ** use of the Next attribute

!Tempo 100
R Q Z10 V1 N0
R Q Z15 V2

A4 H V1 N0
G3 V2

B4 Q V1 N0
F3 V2

C4 Q V1 N0
E3 V2

Here, each pair of lines represents two simultaneous notes. The N0 attribute forces the second line to start at the same time as the first line of each pair. Because of the large intervals, octave numbers (3 and 4) are necessary to override the default octave for these pitches.

## 11.6 Advanced Features

Beyond the simple notation described above, Adagio supports a number of features. (See also the next chapter.)

### 11.6.1 Time Units and Resolution

The default time unit is 10ms (ten milliseconds or one centisecond or 100\(^{th}\) of a second), but it is possible to change the basic unit to 1ms, or 1000\(^{th}\) of a second. The time unit can be specified by:

- !CSEC centisecond time units = 100\(^{th}\)
- !MSEC millisecond time units = 1000\(^{th}\)

The time unit remains in effect until the next !CSEC or !MSEC command.

### 11.6.2 Multiple Notes Per Line

Notes can be separated by commas or semicolons as well as by starting a new line. A comma is equivalent to typing N0 and starting a new line. In other words, the next note after a comma will start at the same time as the note before the comma. In general, use commas to separate the notes of a chord.

A semicolon is equivalent to starting a new line. In general, use semicolons to group notes in a melody. Here is yet another rendition of the Bartok:
*Example 5 ** use of semicolons

!Tempo 100
R Q Z10 V1
A4 H; B Q; C; D H; C; D Q; C; B; A; B; C; D; R

T0 R Q Z15 V2
G3 H; F Q; E; D H; E; D Q; E; F; G; F; E; D; R

This example is similar to Example 2, except semicolons are used. Note how semicolons make the two lines of music stand out. The next example is similar to Example 4, except commas are used and four bars are notated. The music below is treated as a sequence of 2-note chords, with each chord on a separate line:

*Example 6 ** use of commas

!Tempo 100
R Q Z10 V1, R Q Z15 V2
A4 H V1, G3 V2
B4 Q V1, F3 V2
C4 V1, E3 V2
D4 H V1, D3 V2
C4 V1, E3 V2
D4 Q V1, D3 V2
C4 V1, E3 V2
B4 V1, F3 V2
A4 V1, G3 V2
B4 V1, F3 V2
C4 V1, E3 V2
D4 V1, D3 V2
R

### 11.6.3 Control Change Commands

Any control change can be specified using the syntax "~n (v) ", where n is the controller number (0 - 127), and v is the value. In addition, Adagio has some special syntax for some of the commonly used control changes (note that Pitch bend, Aftertouch, and MIDI Program Change are technically not MIDI control changes but have their own special message format and status bytes):

- **K** Portamento switch
- **M** Modulation wheel
- **O** Aftertouch
- **X** Volume
- **Y** Pitch Bend
- **Z** Program Change

The letter listed beside each control function is the Adagio command letter. For example, M23 is the command for setting the modulation wheel to 23. Except for pitch bend, the portamento switch, and MIDI Program Change, all values range from 0 to 127. Pitch bend is “off” or centered at 128, and has a range from 0 to 255 (MIDI allows for more precision, but Adagio does not). Turn on portamento with K127 and off with K0. Programs are numbered 1 to 128 to correspond to synthesizer displays.
About volume: Midi volume is just a control, and the Midi standard does not say what it means. Typically it does what the volume pedal does; that is, it scales the amplitude in a continuously changeable fashion. In contrast, Midi velocity, which is controlled by the L (loudness) attribute, is part of a Midi note-on command and is fixed for the duration of the note. Typically, these two ways of controlling loudness and amplitude operate independently. In some low-cost synthesizers the numbers seem to be added together internally and volume changes are ignored after the note starts.

About pitch bend: Midi pitch bend is a number from 0 to 16383, where 8192 is the center position. To convert to Midi, Adagio simply multiplies your number by 64, giving values from 0 to 16320. Note that Y128 translates exactly to 8192. The meaning of pitch bend depends upon your synthesizer and its setting. Most synthesizers let you specify a "pitch bend range." A range of one semitone means that Y255 will produce a bend of approximately one semitone up, and Y0 will bend one semitone down. If the range is 12 semitones, then the same Y255 will bend an octave. Typically, pitch bend is exponential, so each increment in the pitch bend value will bend an equal number of cents in pitch.

Control changes can be part of a note specification or independent. In the following example, a middle C is played with a modulation wheel setting of 50 and a pitch bend of 120. Then, at 10 unit intervals, the pitch bend is decreased by 10. The last line sets the portamento time (controller 5) to 80:

*Example 7
C4 LMF M50 Y120 U100 N10
Y110 N10; Y100 N10; Y90 N10; Y80 N10
Y70 N10; Y60 N10; Y50 N10
\~5(80)

See Section 11.4 on page 149 for rules on whether or not a command will play a note.

11.6.4 Multiple Tempi

Writing a piece with multiple tempi requires no new commands; you just have to be clever in the use of Tempo and Time. The following plays a 7 note diatonic scale on voice 1, and a 12 note chromatic scale on voice 2:

*Example 8 ** multiple tempi
!TEMPO 70
V1 C4; D; E; F; G; A; B
T0 R N0

!TEMPO 120
V2 C4; CS; D; DS; E; FS; G; GS; A; AS; B

!TEMPO 100
V1 C5, V2 C5

The third line plays the 7-note diatonic scale on voice 1. The next line contains the tricky part: notice that the time is set back to zero, there is a rest, and a next (N) attribute is used to specify that the next default time will be at the same time as the current one. This is tricky because a !TEMPO command cannot have a time (T0) attribute, and a T0 by itself would create a note with a duration. T0 R N0 says: "go to time 0, do not play a note, and do not advance the time before the next command". Thus, the time of the !TEMPO 120 command is zero. After the 12 note scale, the tempo is changed to 100 and a final note is played on each voice. A little arithmetic will show that 7 notes at tempo 70 and 12 notes at tempo 120 each take 6 seconds, so the final notes (C5) of each scale will happen at the same time.
11.6.5 MIDI Synchronization

The Adagio program (but not Nyquist) can synchronize with external devices using MIDI real time messages. Thus, Adagio has a !CLOCK command. This command is currently of no use to Nyquist users but is documented here for completeness (it’s part of the language syntax even if it does not do anything).

Since Adagio supports multiple tempi, and Midi clock is based on beats, it is necessary to be explicit in the score about where the clock should start and what is the duration of a quarter note. The !CLOCK command in Adagio turns on a 24 pulse-per-quarter (PPQ) clock at the current tempo and time:

```
!TEMPO 100
!CLOCK
```

A !CLOCK command must also be inserted for each tempo change that is to be reflected in the Midi clock. Typically, each !TEMPO command will be followed by a !CLOCK command.

Clock commands and thus tempo changes can take place at arbitrary times. It is assumed that tempo changes on an exact 24th of a beat subdivision (for example, exactly on a beat). If not, the tempo change will take place on the nearest exact 24th of a beat subdivision. This may be earlier or later than the requested time.

11.6.6 System Exclusive Messages

Adagio has a definition facility that makes it possible to send system exclusive parameters. Often, there are parameters on Midi synthesizers that can only be controlled by system exclusive messages. Examples include the FM ratio and LFO rate on a DX7 synthesizer. The following example defines a macro for the DX7 LFO rate and then shows how the macro is used to set the LFO rate for a B-flat whole note in the score. The macro definition is given in hexadecimal, except v is replaced by the channel (voice) and %1 is replaced by the first parameter. A macro is invoked by writing "~" followed by the macro name and a list of parameters:

```
!DEF LFO F0 43 0v 01 09 %1 F7
Bf5 W ~LFO(25)
```

In general, the !DEF command can define any single MIDI message including a system exclusive message. The message must be complete (including the status byte), and each !DEF must correspond to just one message. The symbol following !DEF can be any name consisting of alphanumeric characters. Following the name is a hexadecimal string (with optional spaces), all on one line. Embedded in the string may be the following special characters:

\[ v \]

Insert the 4-bit voice (MIDI channel) number. If v occurs in the place of a high-order hexadecimal digit, replace v with 0v so that the channel number is always placed in the low-order 4 bits of a data byte. In other words, v is padded if necessary to fall into the low-order bits.

\[ %n \]

Insert a data byte with the low-order 7 bits of parameter number n. Parameters are numbered 1 through 9. If the parameter value is greater than 127, the high-order bits are discarded.

\[ ^n \]

Insert a data byte with bits 7 through 13 of parameter number n. In other words, shift the value right 7 places then clear all but the first 7 bits. Note that 14-bit numbers can be encoded by referencing the same parameter twice; for example, %4^4 will insert the low-order followed by the high-order parts of parameter 4 into two successive data bytes.
Parameters are separated by commas, but there may be no spaces. The maximum number of parameters allowed is 9. Here is an example of definitions to send a full-resolution pitch bend command and to send a system exclusive command to change a DX7 parameter:

* Define macro for pitch bend commands:

```plaintext
!DEF bend Ev %1 ˆ1
```

A `bend(8192)` ** 8192 is "pitch bend off"

* Change the LFO SPEED:

* SYSEX = F0, Yamaha = 43, Substatus/Channel = 1v,
* Group# = 01, Parameter# = 9, Data = 0-99, EOX = F7

```plaintext
!DEF lfospeed F0 43 1v 01 09 %1 F7
```

* now use the definitions:

```plaintext
G4 ~bend(7567) N40
~lfospeed(30) N35
```

### 11.6.7 Control Ramps

The !RAMP command can specify a smooth control change from one value to another. It consists of a specification of the starting and ending values of some control change, a duration specifying how often to send a new value, and a duration specifying the total length of the ramp.

```plaintext
!RAMP X10 X100 Q W2
!RAMP ~23(10) ~23(50) U20 W
!RAMP ~lfo(15) ~lfo(35) U10
```

The first line says to ramp the volume control (controller number 7) from 10 to 100, changing at each quarter note for the duration of two whole notes. The second line says to ramp controller number 23 from value 10 to value 50, sending a new control change message every 20 time units. The overall duration of the ramp should be equivalent to a whole note (W). As shown in the third line, even system exclusive messages controlled by parameters can be specified. If the system exclusive message has more than one parameter, only one parameter may be "ramped"; the others must remain the same. For example, the following would ramp the second parameter:

```plaintext
!RAMP ~mysysex(4,23,75) ~mysysex(4,100,75) U10 W
```

A rather curious and extreme use of macros and ramps is illustrated in the following example. The noteon macro starts a note, and noteoff ends it. Ramps can now be used to emit a series of notes with changing pitches or velocities. Since Adagio has no idea that these macros are turning on notes, it is up to the programmer to turn them off!

---

2My TX816 Owner's Manual gives an incorrect format for the change parameter sysex command (according to the manual, there is no data in the message!) I am assuming that the data should be the last byte before the EOX and that there is no byte count. If you are reading this, assume that I have not tested this guess, nor have I tested this example.
11.6.8 The !End Command

The special command !END marks the end of a score. Everything beyond that is ignored, for example:

* this is a score
C; D; E; F; G W
!END
since the score has ended, this text will be ignored

11.6.9 Calling C Routines

It is possible to call C routines from within Adagio scores when using specially linked versions, but this feature is disabled in Nyquist. The syntax is described here for completeness.

The !CALL command calls a C routine that can in turn invoke a complex sequence of operations. Below is a call to a trill routine, which is a standard routine in Adagio. The parameters are the base pitch of the trill, the total duration of the trill, the interval in semitones, the duration of each note of the trill, and the loudness. Notice that both numbers and Adagio notation can be used as parameters:

!CALL trill(A5,W,2,S,Lmf) T278 V1

The parameter list should have no spaces, and parameters are separated by commas. Following the close parenthesis, you may specify other attributes such as the starting time and voice as shown in the example above.

A parameter may be an Adagio pitch specification, an Adagio duration, an Adagio loudness, a number, or an ASCII character within single quotes, e.g. 'a' is equivalent to 97 because 97 is the decimal encoding of "a" in ASCII.

The !CALL may be followed by a limited set of attributes. These are time (T), voice (V), and next time (N). The !CALL is made at the current time if no time is specified, and the time of the next adagio command is the time of the !CALL unless a next time is specified. In other words, the default is N0.

11.6.10 Setting C Variables

In addition to calling C routines, there is another way in which scores can communicate with C. As with !CALL, specific C code must be linked before these commands can be used, and this is not supported in Nyquist. The !SETI command sets an integer variable to a value, and the !SETV command sets an element of an integer array. For example, the next line sets the variable delay to 200 and sets transposition[5] to -4 at time 200:

!SETI delay 200
!SETV transposition 5 -4 T200
As with the \texttt{!CALL} command, these commands perform their operations at particular times according to their place in the Adagio score. This makes it very easy to implement time-varying parameters that control various aspects of an interactive music system.
12  Linear Prediction Analysis and Synthesis

Nyquist provides functions to perform Linear Prediction Coding (LPC) analysis and synthesis. In simple terms, LPC analysis assumes that a sound is the result of an all-pole filter applied to a source with a flat spectrum. LPC is good for characterizing the general spectral shape of a signal, which may be time-varying as in speech sounds. For synthesis, any source can be filtered, allowing the general spectral shape of one signal (used in analysis) to be applied to any source (used in synthesis). A popular effect is to give vowel-like spectra to musical tones, creating an artificial (or sometimes natural) singing voice.

Examples of LPC analysis and synthesis can be found in the file demos/lpc_tutorial.htm, which is part of the standard Nyquist release.

As with FFT processing, LPC analysis takes a sound as input and returns a stream of frames. Frames are returned from an object using the :next selector just as with FFT frames. An LPC frame is a list consisting of: RMS1, the energy of the input signal, RMS2, the energy of the residual signal, ERR, the square root of RMS1/RMS2, and FILTER-COEFS, an array of filter coefficients. To make code more readable and to avoid code dependence on the exact format of a frame, the functions lpc-frame-rms1, lpc-frame-rms2, lpc-frame-err, and lpc-frame-filter-coefs can be applied to a frame to obtain the respective fields.

The \( z \) transform of the filter is \( H(z) = \frac{1}{A(z)} \), where \( A(z) \) is a polynomial of the form \( A(z) = 1 + a_1z + a_2z^2 + \ldots + a_pz^p \). The FILTER-COEFS array has the form \((a_p, a_{p-1}, \ldots, a_3, a_2, a_1)\).

The file lpc.lsp defines some useful classes and functions. The file is not automatically loaded with Nyquist, so you must execute `(load "lpc")` before using them.

12.1  LPC Classes and Functions

```
(make-lpanal-iterator sound framedur skiptime npoles) [SAL]

(make-lpanal-iterator sound framedur skiptime npoles) [LISP]
```

Makes an iterator object, an instance of lpanal-class, that returns LPC frames from successive frames of samples in sound. The duration (in seconds) of each frame is given by framedur, a FLONUM. The skip size (in seconds) between successive frames is given by skiptime, a FLONUM. Typical values for framedur and skiptime are 0.08 and 0.04, giving 25 frames per second and a 50% frame overlap. The number of poles is given by npoles, a FIXNUM. The result is an object that responds to the :next selector by returning a frame as described above. NIL is returned when sound terminates. (Note that one or more of the last analysis windows may be padded with zeros. NIL is only returned when the corresponding window would begin after the termination time of the sound.)
Another way to get LPC frames is to read them from a file. This function opens an ASCII file containing LPC frames and creates an iterator object, an instance of class `lpc-file-class` to access them. Create a file using `save-lpc-file` (see below).

```lisp
(save-lpc-file lpc-iterator filename)
```

Create a file containing LPC frames. This file can be read by `make-lpc-file-iterator` (see above).

```lisp
(make-lpc-file-iterator filename)
```

Print values of LPC frames from an LPC iterator object. The object is `lpc-iterator`, which is typically an instance of `lpanal-class` or `lpc-file-class`. Frames are numbered from zero, and only files starting at `iniframe` (a `FIXNUM`) and ending before `endframe` (also a `FIXNUM`) are printed. By default, only the values for `RMS1`, `RMS2`, and `ERR` are printed, but if optional parameter `poles?` is non-NIL, then the LPC coefficients are also printed.

```lisp
(show-lpc-data lpc-iterator iniframe endframe [poles?])
```

A single LPC frame defines a filter. Use `allpoles-from-lpc` to apply this filter to `snd`, a `SOUND`. To obtain `lpc-frame`, a `LIST` containing an LPC frame, either send `:next` to an LPC iterator, or use `nth-frame` (see below). The result is a `SOUND` whose duration is the same as that of `snd`.

```lisp
(allpoles-from-lpc snd lpc-frame)
```

Implements a time-varying all-pole filter controlled by a sequence of LPC frames from an iterator. The `SOUND` to be filtered is `snd`, and the source of LPC frames is `lpc-iterator`, typically an instance of `lpanal-class` or `lpc-file-class`. The frame period (in seconds) is given by `skiptime` (a `FLONUM`). This number does not have to agree with the `skiptime` used to analyze the frames. (Greater values will cause the filter evolution slow down, and smaller values will cause it to speed up.) The result is a `SOUND`. The duration of the result is the minimum of the duration of `snd` and that of the sequence of frames.

```lisp
(lpreson snd lpc-iterator skiptime)
```

Get the energy of the input signal from a frame.

```lisp
(lpc-frame-rms1 frame)
```

Get the energy of the residual from a frame.

```lisp
(lpc-frame-rms2 frame)
```

Get the square root of \(RMS1/RMS2\) from a frame.

```lisp
(lpc-frame-filter-coefs frame)
```

Get the filter coefficients from a frame.

### 12.2 Low-level LPC Functions

The lowest-level Nyquist functions for LPC are
• `snd-lpanal` for analysis,

• `snd-allpoles`, an all-pole filter with fixed coefficients, and

• `snd-lpreson`, an all-pole filter that takes frames from an LPC iterator.

```
snd-lpanal(samps, npoles) [SAL]
snd-lpanal samps npoles) [LISP]
```

Compute an LPC frame with `npoles` (a FIXNUM) poles from an ARRAY of samples (FLONUMS). Note that `snd-fetch-array` can be used to fetch a sequence of frames from a sound. Ordinarily, you should not use this function. Use `make-lpanal-iterator` instead.

```
snd-allpoles(snd, lpc-coefs, gain) [SAL]
(snd-allpoles snd lpc-coefs gain) [LISP]
```

A fixed all-pole filter. The input is `snd`, a SOUND. The filter coefficients are given by `lpc-coefs` (an ARRAY), and the filter gain is given by `gain`, a FLONUM. The result is a SOUND whose duration matches that of `snd`. Ordinarily, you should use `allpoles-from-lpc` instead (see above).

```
snd-lpreson(snd, lpc-iterator, skiptime) [SAL]
(snd-lpreson snd lpc-iterator skiptime) [LISP]
```

This function is identical to `lpreson` (see above).
13 Developing and Debugging in Nyquist

There are a number of tools, functions, and techniques that can help to debug Nyquist programs. Since these are described in many places throughout this manual, this chapter brings together many suggestions and techniques for developing code and debugging. You really should read this chapter before you spend too much time with Nyquist. Many problems that you will certainly run into are addressed here.

13.1 Debugging

Probably the most important debugging tool is the backtrace. There are two kinds of backtrace: one for SAL, and one for Lisp.

SAL mode is actually just an XLISP function (sal) that reads input and evaluates it. When SAL encounters an error, it normally prints a trace of the SAL stack (all the active functions written in SAL), exists the current command, and reads the next command.

If you call XLISP functions from SAL, including most Nyquist sound processing functions, and an error occurs within these XLISP functions, you will only see the SAL function that called the XLISP functions listed in the stack trace. Sometimes you need more details.

When Nyquist encounters an error when it is not running SAL, it normally suspends execution and prints an error message. To find out where in the program the error occurred and how you got there, start by typing (bt). This will print out the last several function calls and their arguments, which is usually sufficient to see what is going on.

In order for (bt) to work, you must have a couple of global variables set: *tracenable* is ordinarily set to NIL. If it is true, then a backtrace is automatically printed when an error occurs; *breakenable* must be set to T, as it enables the execution to be suspended when an error is encountered. If *breakenable* is NIL (false), then execution stops when an error occurs but the stack is not saved and you cannot get a backtrace. Finally, bt is just a macro to save typing. The actual backtrace function is baktrace, which takes an integer argument telling how many levels to print. All of these things are set up by default when you start Nyquist.

To get this XLISP backtrace behavior when SAL encounters an error, you need to have *breakenable* set while SAL is running. The best way to do this is to run within the NyquistIDE program, open the Preferences dialog, and choose the desired settings, e.g. "Enable XLISP break on SAL error."

Since Nyquist sounds are executed with a lazy evaluation scheme, some errors are encountered when samples are being generated. In this case, it may not be clear which expression is in error. Sometimes, it is best to explore a function or set of functions by examining intermediate results. Any expression that yields a sound can be assigned to a variable and examined using one or more of: s-plot, snd-print-tree, and of course play. The snd-print-tree function
prints a lot of detail about the inner representation of the sound. Keep in mind that if you assign a sound to a global variable and then look at the samples (e.g. with play or s-plot), the samples will be retained in memory. At 4 bytes per sample, a big sound may use all of your memory and cause a crash.

Another technique is to use low sample rates so that it is easier to plot results or look at samples directly. The calls:

```
set-sound-srate(100)
set-control-srate(100)
```

set the default sample rates to 100, which is too slow for audio, but useful for examining programs and results. The function

```
snd-samples(sound, limit)
```

will convert up to `limit` samples from `sound` into a Lisp array. This is another way to look at results in detail.

The `trace` function is sometimes useful. It prints the name of a function and its arguments every time the function is called, and the result is printed when the function exits. To trace the osc function, type:

```
trace(osc)
```

and to stop tracing, type `untrace(osc)`.

If a variable needs a value or a function is undefined, and if `*breakenable*` was set, you will get a prompt where you can fix the error (by setting the variable or loading the function definition) and keep going. At the debug (or break) prompt, your input must be in XLISP, not SAL syntax. Use `(co)`, short for `(continue)` to reevaluate the variable or function and continue execution.

When you finish debugging a particular call, you can "pop" up to the top level by typing `(top)`, a short name for `(top-level)`. There is a button named "Top" in the NyquistIDE that takes you back to the top level (ready to accept XLISP expressions), and another button named "SAL" that puts you back in SAL mode.

### 13.2 Useful Functions

**grindef(name)** [SAL]

```
(grindef name)
```

Prints a formatted listing of a Lisp function. This is often useful to quickly inspect a function without searching for it in source files. Do not forget to quote the name, e.g. `(grindef 'prod)`.

**args(name)** [SAL]

```
(args name)
```

Similar to `grindef`, this function prints the arguments to a function. This may be faster than looking up a function in the documentation if you just need a reminder. For example, `(args 'lp)` prints "(LP S C)," which may help you to remember that the arguments are a sound (S) followed by the cutoff (C) frequency.

The following functions are useful shortcuts that might have been included in XLISP. They are so useful that they are defined as part of Nyquist.

**incf(symbol)** [SAL]

```
(incf symbol)
```

Increment `symbol` by one. This is a macro, and `symbol` can be anything that can be set by `setf`. Typically, `symbol` is a variable: "(incf i)," but `symbol` can also be an array element: "(incf (aref myarray i))."
Decrement symbol by one. (See incf, above.)

Push val onto lis (a Lisp list). This is a macro that is equivalent to writing (in Lisp) (setf lis (cons val lis)).

Remove (pop) the first item from lis (a Lisp list). This is a macro that is equivalent to writing (in Lisp) (setf lis (cdr lis)). Note that the remaining list is returned, not the head of the list that has been popped.

Retrieve the head of the list (i.e. the top of the stack) using first or, equivalently, car.

The following macros are useful control constructs.

A conventional "while" loop. If test is true, evaluate expressions (expr1, expr2, etc.) and repeat. If test is false, return. This expression evaluates to NIL unless the expression (return expr) is evaluated, in which case the value of expr is returned. In SAL, the loop statement is preferred.

A conventional "if-then" statement. If test is true, action is evaluated and returned. Otherwise, NIL is returned. (Use if or cond to implement "if-then-else" and more complex conditional forms.

It is often necessary to load a file only if it has not already been loaded. For example, the pianosyn library loads somewhat slowly, so if some other file already loaded it, it would be good to avoid loading it again. How can you load a file once? Nyquist does not keep track of files that are loaded, but you must be loading a file to define some function, so the idea is to tell Nyquist "I require function from file"; if the function does not yet exist, Nyquist satisfies the requirement by loading the file.

Tests whether fnsymbol, an unquoted function name, is defined. If not, filename, a STRING, is loaded. Normally fnsymbol is a function that will be called from within the current file, and filename is the file that defines fnsymbol. The path, if a STRING, is prepended to filename. If path is t (true), then the directory of the current file is used as the path.

Sometimes it is important to load files relative to the current file. For example, the lib/piano.lsp library loads data files from the lib/piano directory, but how can we find out the full path of lib? The solution is:

Returns the full path name of the file that is currently being loaded (see load). Returns NIL if no file is being loaded.

Finally, there are some helpful math functions:
real-random(from, to) [SAL]
(real-random from to) [LISP]
    Returns a random FLONUM between from and to. (See also rrandom, which is equivalent to (real-random 0 1)).

power(x, y) [SAL]
(power x y) [LISP]
    Returns x raised to the y power.
14 Xmusic and Algorithmic Composition

Several Nyquist libraries offer support for algorithmic composition. Xmusic is a library for generating sequences and patterns of data. Included in Xmusic are:

- pattern objects, used to generate interesting sequences of parameter values,
- random number generators, used to create random sequences from different distributions (contained in the `distributions.lsp` library)
- a standard representation for "note lists" called `scores` and functions to render them as sounds,
- the `score-gen` macro which helps to generate scores from patterns,
- score manipulation functions to select, transform, shift, and perform other operations on scores.

14.1 Xmusic Basics

Xmusic is inspired by and based on Common Music by Rick Taube. Common Music supports MIDI and various other synthesis languages and includes a graphical interface, some visualization tools, and many other features. Common Music runs in Common Lisp and Scheme, but not XLISP (the base language for Nyquist).

The Xmusic libraries in Nyquist offer an interesting subset of the tools in Common Music. One important feature of Xmusic is that it is integrated with all of the Nyquist synthesis functions, so you can use Xmusic patterns and scores to control fine details of sound synthesis.

14.2 Xmusic Patterns

Xmusic patterns are objects that generate data streams. For example, the `cycle-class` of objects generate cyclical patterns such as "1 2 3 1 2 3 1 2 3 ...", or "1 2 3 4 3 2 1 2 3 4 ...". Patterns can be used to specify pitch sequences, rhythm, loudness, and other parameters.

Xmusic functions are automatically loaded when you start Nyquist. To use a pattern object, you first create the pattern, e.g.

```
set pitch-source = make-cycle(list(c4, d4, e4, f4))
```
In this example, pitch-source is an object of class cycle-class which inherits from pattern-class.

Because SAL is not an object-oriented language, these classes and their methods are not directly accessible from SAL. Instead, Xmusic defines a functional interface, e.g. make-cycle creates an instance of cycle-class, and the next function, introduced below, retrieves the next value from any instance of pattern-class. Using LISP syntax, you can have full access to the methods of all objects (see the source code in xlsp, but the functional interface described here should be sufficient and it is recommended that you limit your access to this interface.

After creating the pattern, you can access it repeatedly with next to generate data, e.g.

```lisp
play seqrep(i, 13, pluck(next(pitch-source), 0.2))
```

This will create a sequence of notes with the following pitches: c, d, e, f, c, d, e, f, c, d, e, f. If you evaluate this again, the pitch sequence will continue, starting on "d".

It is very important not to confuse the creation of a sequence with its access. Consider this example:

```lisp
play seqrep(i, 13, pluck(next(make-cycle(list(c4, d4, e4, f4))), 0.2))
```

This looks very much like the previous example, but it only repeats notes on middle-C. The reason is that every time pluck is evaluated, make-cycle is called and creates a new pattern object. After the first item of the pattern is extracted with next, the cycle is not used again, and no other items are generated.

To summarize this important point, there are two steps to using a pattern. First, the pattern is created and stored in a variable using setf. Second, the pattern is accessed (multiple times) using next.

### 14.2.1 Nested Patterns

Patterns can be nested, that is, you can write patterns of patterns. In general, the next function does not return patterns. Instead, if the next item in a pattern is a (nested) pattern, next recursively gets the next item of the nested pattern.

While you might expect that each call to next would advance the top-level pattern to the next item, and descend recursively if necessary to the inner-most nesting level, this is not how next works. Instead, next remembers the last top-level item, and if it was a pattern, next continues to generate items from that same inner pattern until the end of the inner pattern’s period is reached. The next paragraph explains the concept of the period.

### 14.2.2 Periods

The data returned by a pattern object is structured into logical groups called periods. You can get an entire period (as a list) by calling next(pattern, t). For example:

```lisp
set pitch-source = make-cycle(list(c4, d4, e4, f4))
print next(pitch-source, t)
```

This prints the list (60 62 64 65), which is one period of the cycle.

You can also get explicit markers that delineate periods by calling send(pattern, :next). In this case, the value returned is either the next item of the pattern, or the symbol +eop+ if the end of a period has been reached. What determines a period? This is up to the specific pattern class, so see the documentation for specifics. You can override the "natural" period using the keyword for:, e.g.
set pitch-source = make-cycle(list(c4, d4, e4, f4), for: 3)
print next(pitch-source, t)
print next(pitch-source, t)

This prints the lists (60 62 64) (65 60 62). Notice that these periods just restructure the stream of items into groups of 3.

Nested patterns are probably easier to understand by example than by specification. Here is a simple nested pattern of cycles:

set cycle-1 = make-cycle({a b c})
set cycle-2 = make-cycle({x y z})
set cycle-3 = make-cycle(list(cycle-1, cycle-2))
exec dotimes(i, 9, format(t, "~A ", next(cycle-3)))

This will print "A B C X Y Z A B C". Notice that the inner-most cycles cycle-1 and cycle-2 generate a period of items before the top-level cycle-3 advances to the next pattern.

### 14.2.3 General Parameters for Creating Pattern objects

Before describing specific pattern classes, there are several optional parameters that apply in the creating of any pattern object. These are:

- **for:**
  The length of a period. This overrides the default by providing a numerical length. The value of this optional parameter may be a pattern that generates a sequence of integers that determine the length of each successive period. A period length may not be negative, but it may be zero.

- **name:**
  A pattern object may be given a name. This is useful if the trace: option is used.

- **trace:**
  If non-null, this optional parameter causes information about the pattern to be printed each time an item is generated from the pattern.

The built-in pattern classes are described in the following section.

### 14.2.4 cycle

The **cycle** class iterates repeatedly through a list of items. For example, two periods of make-cycle({a b c}) would be (A B C) (A B C).

```
make-cycle(items, for: for, name: name, trace: trace) [SAL]
(make-cycle items :for for :name name :trace trace) [LISP]
```

Make a cycle pattern that iterates over items. The default period length is the length of items. (See above for a description of the optional parameters.) If items is a pattern, a period of the pattern becomes the list from which items are generated. The list is replaced every period of the cycle.
14.2.5 line

The line-class is similar to the cycle class, but when it reaches the end of the list of items, it simply repeats the last item in the list. For example, two periods of make-line({a b c}) would be (A B C) (C C C).

```
(make-line items :for for :name name :trace trace) [SAL]
(make-line items :for for :name name :trace trace) [LISP]
```

Make a line pattern that iterates over items. The default period length is the length of items. As with make-cycle, items may be a pattern. (See above for a description of the optional parameters.)

14.2.6 random

The random-class generates items at random from a list. The default selection is uniform random with replacement, but items may be further specified with a weight, a minimum repetition count, and a maximum repetition count. Weights give the relative probability of the selection of the item (with a default weight of one). The minimum count specifies how many times an item, once selected at random, will be repeated. The maximum count specifies the maximum number of times an item can be selected in a row. If an item has been generated \(n\) times in succession, and the maximum is equal to \(n\), then the item is disqualified in the next random selection. Weights (but not currently minima and maxima) can be patterns. The patterns (thus the weights) are recomputed every period.

```
(make-random items :for for :name name :trace trace) [SAL]
(make-random items :for for :name name :trace trace) [LISP]
```

Make a random pattern that selects from items. Any (or all) element(s) of items may be lists of the following form: `(value :weight weight :min mincount :max maxcount)`, where value is the item (or pattern) to be generated, weight is the (optional) relative probability of selecting this item, mincount is the (optional) minimum number of repetitions when this item is selected, and maxcount is the (optional) maximum number of repetitions allowed before selecting some other item. The default period length is the length of items. If items is a pattern, a period from that pattern becomes the list from which random selections are made, and a new list is generated every period.

14.2.7 palindrome

The palindrome-class repeatedly traverses a list forwards and then backwards. For example, two periods of make-palindrome({a b c}) would be (A B C B A) (A B C B A). The elide: keyword parameter controls whether the first and/or last elements are repeated:
make-palindrome({a b c}, elide: nil)
    ;; generates A B C C B A A B C B A ...

make-palindrome({a b c}, elide: t)
    ;; generates A B C B A B C B ...

make-palindrome({a b c}, elide: :first)
    ;; generates A B C C B A B C C B ... 

make-palindrome({a b c}, elide: :last)
    ;; generates A B C B A A B C B A ...

make-palindrome(items, elide: elide, for: for, name: name, trace: trace)[SAL]
(make-palindrome items :elide elide :for for :name name :trace trace)[LISP]

Generate items from list alternating in-order and reverse-order sequencing. The keyword parameter `elide` can have the values :first, :last, t, or nil to control repetition of the first and last elements. The `elide` parameter can also be a pattern, in which case it is evaluated every period. One period is one complete forward and backward traversal of the list. If `items` is a pattern, a period from that pattern becomes the list from which random selections are made, and a new list is generated every period.

14.2.8 heap

The `heap-class` selects items in random order from a list without replacement, which means that all items are generated once before any item is repeated. For example, two periods of `make-heap({a b c})` might be (C A B) (B A C). Normally, repetitions can occur even if all list elements are distinct. This happens when the last element of a period is chosen first in the next period. To avoid repetitions, the `max:` keyword argument can be set to 1. The `max:` keyword only controls repetitions from the end of one period to the beginning of the next. If the list contains more than one copy of the same value, it may be repeated within a period regardless of the value of `max:`.

make-heap(items, for: for, max: max, name: name, trace: trace)[SAL]
(make-heap items :for for :max max :name name :trace trace)[LISP]

Generate items randomly from list without replacement. If `max` is 1, the first element of a new period will not be the same as the last element of the previous period, avoiding repetition. The default value of `max` is 2, meaning repetition is allowed. The period length is the length of `items`. If `items` is a pattern, a period from that pattern becomes the list from which random selections are made, and a new list is generated every period.

14.2.9 accumulation

The `accumulation-class` takes a list of values and returns the first, followed by the first two, followed by the first three, etc. In other words, for each list item, return all items from the first through the item. For example, if the list is (A B C), each generated period is (A A B A B C).
make-accumulation(items, name: name, trace: trace) [SAL]
(make-accumulation items :name name :trace trace) [LISP]

Return the prefixes of items, e.g. the first element followed by the first and second elements, then the first three, until the entire list is returned. The period length is \((n^2 + n) / 2\) where \(n\) is the length of items. If items is a pattern, a period from that pattern becomes the list from which items are generated, and a new list is generated every period. Note that this is similar in name but different from make-accumulate.

14.2.10 copier

The copier-class makes copies of periods from a sub-pattern. For example, three periods of make-copier(make-cycle({a b c}, for: 1), repeat: 2, merge: t) would be (A A) (B B) (C C). Note that entire periods (not individual items) are repeated, so in this example the for: keyword was used to force periods to be of length one so that each item is repeated by the repeat: count.

make-copier(sub-pattern, repeat: repeat, merge: merge, for: for, name: name, trace: trace) [SAL]
(make-copier sub-pattern :repeat repeat :merge merge :for for :name name :trace trace) [LISP]

Generate a period from sub-pattern and repeat it repeat times. If merge is false (the default), each repetition of a period from sub-pattern results in a period by default. If merge is true (non-null), then all repeat repetitions of the period are merged into one result period by default. If the for: keyword is used, the same items are generated, but the items are grouped into periods determined by the for: parameter. If the for: parameter is a pattern, it is evaluated every result period. The repeat and merge values may be patterns that return a repeat count and a boolean value, respectively. If so, these patterns are evaluated initially and after each repeat copies are made (independent of the for: keyword parameter, if any). The repeat value returned by a pattern can also be negative. A negative number indicates how many periods of sub-pattern to skip. After skipping these patterns, new repeat and merge values are generated.

14.2.11 accumulate

The accumulate-class forms the sum of numbers returned by another pattern. For example, each period of make-accumulate(make-cycle({1 2 -3})) is (1 3 0). The default output period length is the length of the input period.

make-accumulate(sub-pattern, for: for, max: maximum, min: minimum, name: name, trace: trace) [SAL]
(make-accumulate sub-pattern :for for :max maximum :min minimum :name name :trace trace) [LISP]

Keep a running sum of numbers generated by sub-pattern. The default period lengths match the period lengths from sub-pattern. If maximum (a pattern or a number) is specified, and the running sum exceeds maximum, the running sum is reset to maximum. If minimum (a pattern or a number) is specified, and the running sum falls below minimum, the running sum is reset to minimum. If minimum is greater than maximum, the running sum will be set to one of the two values. Note that this is similar in name but not in function to make-accumulation.
14.2.12 sum

The sum-class forms the sum of numbers, one from each of two other patterns. For example, each period of make-sum(make-cycle({1 2 3}), make-cycle({4 5 6})) is (5 7 9). The default output period length is the length of the input period of the first argument. Therefore, the first argument must be a pattern, but the second argument can be a pattern or a number.

\[
\text{make-sum}(x, y, \text{for: } \text{for}, \text{name: } \text{name}, \text{trace: } \text{trace}) \quad \text{[SAL]}
\]
\[
\text{(make-sum x y :for for :name name :trace trace)} \quad \text{[LISP]}
\]

Form sums of items (which must be numbers) from pattern \(x\) and pattern or number \(y\). The default period lengths match the period lengths from \(x\).

14.2.13 product

The product-class forms the product of numbers, one from each of two other patterns. For example, each period of make-product(make-cycle({1 2 3}), make-cycle({4 5 6})) is (4 10 18). The default output period length is the length of the input period of the first argument. Therefore, the first argument must be a pattern, but the second argument can be a pattern or a number.

\[
\text{make-product}(x, y, \text{for: } \text{for}, \text{name: } \text{name}, \text{trace: } \text{trace}) \quad \text{[SAL]}
\]
\[
\text{(make-product x y :for for :name name :trace trace)} \quad \text{[LISP]}
\]

Form products of items (which must be numbers) from pattern \(x\) and pattern or number \(y\). The default period lengths match the period lengths from \(x\).

14.2.14 eval

The eval-class evaluates an expression to produce each output item. The default output period length is 1.

\[
\text{make-eval}(\text{expr}, \text{for: } \text{for}, \text{name: } \text{name}, \text{trace: } \text{trace}) \quad \text{[SAL]}
\]
\[
\text{(make-eval expr :for for :name name :trace trace)} \quad \text{[LISP]}
\]

Evaluate \(\text{expr}\) to generate each item. If \(\text{expr}\) is a pattern, each item is generated by getting the next item from \(\text{expr}\) and evaluating it.

14.2.15 length

The length-class generates periods of a specified length from another pattern. This is similar to using the for: keyword, but for many patterns, the for: parameter alters the points at which other patterns are generated. For example, if the palindrome pattern has an \(\text{elide: }\) pattern parameter, the value will be computed every period. If there is also a for: parameter with a value of 2, then \(\text{elide:}\) will be recomputed every 2 items. In contrast, if the palindrome (without a for: parameter) is embedded in a length pattern with a length of 2, then the periods will all be of length 2, but the items will come from default periods of the palindrome, and therefore the \(\text{elide:}\) values will be recomputed at the beginnings of default palindrome periods.
make-length(pattern, length-pattern, name: name, trace: trace) [SAL]
(make-length pattern length-pattern :name name :trace trace) [LISP]

Make a pattern of class length-class that regroups items generated by a pattern according to pattern lengths given by length-pattern. Note that length-pattern is not optional: There is no default pattern length and no for: keyword.

14.2.16 window

The window-class groups items from another pattern by using a sliding window. If the skip value is 1, each output period is formed by dropping the first item of the previous periods and appending the next item from the pattern. The skip value and the output period length can change every period. For a simple example, if the period length is 3 and the skip value is 1, and the input pattern generates the sequence A, B, C, ... , then the output periods will be (A B C), (B C D), (C D E), (D E F), ....

make-window(pattern, length-pattern, skip-pattern, name: name, trace: trace) [SAL]
(make-window pattern length-pattern skip-pattern :name name :trace trace) [LISP]

Make a pattern of class window-class that regroups items generated by a pattern according to pattern lengths given by length-pattern and where the period advances by the number of items given by skip-pattern. Note that length-pattern is not optional: There is no default pattern length and no for: keyword.

14.2.17 markov

The markov-class generates items from a Markov model. A Markov model generates a sequence of states according to rules which specify possible future states given the most recent states in the past. For example, states might be pitches, and each pitch might lead to a choice of pitches for the next state. In the markov-class, states can be either symbols or numbers, but not arbitrary values or patterns. This makes it easier to specify rules. However, symbols can be mapped to arbitrary values including pattern objects, and these become the actual generated items. By default, all future states are weighted equally, but weights may be associated with future states. A Markov model must be initialized with a sequence of past states using the past: keyword. The most common form of Markov model is a "first order Markov model" in which the future item depends only upon one past item. However, higher order models where the future items depend on two or more past items are possible. A "zero-order" Markov model, which depends on no past states, is essentially equivalent to the random pattern. As an example of a first-order Markov pattern, two periods of make-markov({{a -> b c} {b -> c} {c -> a}}, past: {a}) might be (C A C) (A B C).

make-markov(rules, past: past, produces: produces, for: for, name: name, trace: trace) [SAL]
(make-markov rules past: produces produces :for for :name name :trace trace) [LISP]

Generate a sequence of items from a Markov process. The rules parameter is a list of rules where each rule has the form: (prev1 prev2 ... prevn -> next1 next2 ... nextn) where prev1 through prevn represent a sequence of most recent (past) states. The symbol * is treated specially: it matches any previous state. If prev1 through prevn (which may be just one state as in the example above) match the previously generated states, this rule applies. Note that every rule must specify the same number of previous states; this number is known as the order of the Markov model. The first rule in rules that applies is used to select the next state. If no rule applies, the next state is NIL (which is a valid state that can be used in rules). Assuming a rule applies,
the list of possible next states is specified by next1 through nextn. Notice that these are alternative choices for the next state, not a sequence of future states, and each rule can have any number of choices. Each choice may be the state itself (a symbol or a number), or the choice may be a list consisting of the state and a weight. The weight may be given by a pattern, in which case the next item of the pattern is obtained every time the rule is applied. For example, this rules says that if the previous states were A and B, the next state can be A with a weight of 0.5 or C with an implied weight of 1: (A B -> (A 0.5) C). The default length of the period is the length of rules. The past parameter must be provided. It is a list of states whose length matches the order of the Markov model. The keyword parameter produces may be used to map from state symbols or numbers to other values or patterns. The parameter is a list of alternating symbols and values. For example, to map A to 69 and B to 71, use list(quote(a), 69, quote(b), 71). You can also map symbols to patterns, for example list(quote(a), make-cycle({57 69}), quote(b), make-random({59 71})). The next item of the pattern is is generated each time the Markov model generates the corresponding state. Finally, the produces keyword can be :eval, which means to evaluate the Markov model state. This could be useful if states are Nyquist global variables such as C4, CS4, D4, ... , which evaluate to numerical values (60, 61, 62, ....

markov-create-rules(sequence, order [, generalize]) [SAL]
(markov-create-rules sequence order [generalize]) [LISP]

Generate a set of rules suitable for the make-markov function. The sequence is a "typical" sequence of states, and order is the order of the Markov model. It is often the case that a sample sequence will not have a transition from the last state to any other state, so the generated Markov model can reach a "dead end" where no rule applies. This might lead to an infinite stream of NIL's. To avoid this, the optional parameter generalize can be set to t (true), indicating that there should be a fallback rule that matches any previous states and whose future states are weighted according to their frequency in sequence. For example, if sequence contains 5 A's, 5 B's and 10 G's, the default rule will be ( * -> (A 5) (B 5) (G 10) ). This rule will be appended to the end so it will only apply if no other rule does.

14.3 Random Number Generators

The distributions.lsp library implements random number generators that return random values with various probability distributions. Without this library, you can generate random numbers with uniform distributions. In a uniform distribution, all values are equally likely. To generate a random integer in some range, use random. To generate a real number (FLONUM) in some range, use real-random (or rrandom if the range is 0-1). But there are other interesting distributions. For example, the Gaussian distribution is often used to model real-world errors and fluctuations where values are clustered around some central value and large deviations are more unlikely than small ones. See Dennis Lorrain, "A Panoply of Stochastic 'Canons'," Computer Music Journal vol. 4, no. 1, 1980, pp. 53-81. Further discussion and examples can be found in demos/probability_distributions.htm.

In most of the random number generators described below, there are optional parameters to indicate a maximum and/or minimum value. These can be used to truncate the distribution. For example, if you basically want a Gaussian distribution, but you never want a value greater than 5, you can specify 5 as the maximum value. The upper and lower bounds are implemented simply by drawing a random number from the full distribution repeatedly until a number falling into the desired range is obtained. Therefore, if you select an acceptable range that is unlikely, it may take Nyquist a long time to find each acceptable random number. The intended use of the upper and lower bounds is to weed out values that are already fairly unlikely.
Figure 14.1: The Linear Distribution, \( g = 1 \).

\texttt{linear-dist(g)} [SAL]

\texttt{(linear-dist g)} [LISP]

Return a \texttt{FLONUM} value from a linear distribution, where the probability of a value decreases linearly from zero to \( g \) which must be greater than zero. (See Figure 14.1.) The linear distribution is useful for generating for generating time and pitch intervals.

\texttt{exponential-dist(delta [, high])} [SAL]

\texttt{(exponential-dist delta [high])} [LISP]

Return a \texttt{FLONUM} value from an exponential distribution. The initial downward slope is steeper with larger values of \( delta \), which must be greater than zero. (See Figure 14.2. The optional \texttt{high} parameter puts an artificial upper bound on the return value. The exponential distribution generates values greater than 0, and can be used to generate time intervals. Natural random intervals such as the time intervals between the release of atomic particles or the passing of yellow volkswagons in traffic have exponential distributions. The exponential distribution is memory-less: knowing that a random number from this distribution is greater than some value (e.g. a note duration is at least 1 second) tells you nothing new about how soon the note will end. This is a continuous distribution, but \texttt{geometric-dist} (described below) implements the discrete form.

\texttt{gamma-dist(nu [, high])} [SAL]

\texttt{(gamma-dist nu [high])} [LISP]

Return a \texttt{FLONUM} value from a Gamma distribution. The value is greater than zero, has a mean of \( nu \) (a \texttt{FIXNUM} greater than zero), and a mode (peak) of around \( nu - 1 \). The optional \texttt{high} parameter puts an artificial upper bound on the return value.

\texttt{bilateral-exponential-dist(xmu, tau [, low, high])} [SAL]

\texttt{(bilateral-exponential-dist xmu tau [low high])} [LISP]

Returns a \texttt{FLONUM} value from a bilateral exponential distribution, where \( xmu \) is the center of the double exponential and \( tau \) controls the spread of the distribution. A larger \( tau \) gives a wider distribution (greater variance), and \( tau \) must be greater than zero. The \texttt{low} and \texttt{high} parameters give optional artificial bounds on the minimum and maximum output values, respectively. This distribution is similar to the exponential, except it is centered at 0 and can output negative values as well. Like the exponential, it can be used to generate time intervals; however, it might be necessary to add a lower bound so as not to compute a negative time interval.
Figure 14.2: The Exponential Distribution, \( \delta = 1 \).

Figure 14.3: The Gamma Distribution, \( \nu = 4 \).
cauchy-dist(\( \tau \), low, high) [SAL]

(cauchy-dist \( \tau \) [low high]) [LISP]

Returns a FLONUM from the Cauchy distribution, a symmetric distribution with a high peak at zero and a width (variance) that increases with parameter \( \tau \), which must be greater than zero. The \( \text{low} \) and \( \text{high} \) parameters give optional artificial bounds on the minimum and maximum output values, respectively.

hyperbolic-cosine-dist([low, high]) [SAL]

(hyperbolic-cosine-dist [low high]) [LISP]

Returns a FLONUM value from the hyperbolic cosine distribution, a symmetric distribution with its peak at zero. The \( \text{low} \) and \( \text{high} \) parameters give optional artificial bounds on the minimum and maximum output values, respectively.
logistic-dist(alpha, beta [, low, high]) [SAL]
(logistic-dist alpha beta [low high]) [LISP]

Returns a FLONUM value from the logistic distribution, which is symmetric about the mean. The \textit{alpha} parameter primarily affects dispersion (variance), with larger values resulting in values closer to the mean (less variance), and the \textit{beta} parameter primarily influences the mean. The \textit{low} and \textit{high} parameters give optional artificial bounds on the minimum and maximum output values, respectively.

arc-sine-dist() [SAL]
(arc-sine-dist) [LISP]

Returns a FLONUM value from the arc sine distribution, which outputs values between 0 and 1. It is symmetric about the mean of 1/2, but is more likely to generate values closer to 0 and 1.
Figure 14.8: The Arc Sine Distribution.

Figure 14.9: The Gauss-Laplace (Gaussian) Distribution, $xmu = 0, sigma = 1$.

gaussian-dist($xmu, sigma [, low, high]$) [SAL]
(gaussian-dist $xmu sigma [low high]$) [LISP]
Returns a FLONUM value from the Gaussian or Gauss-Laplace distribution, a linear function of the normal distribution. It is symmetric about the mean of $xmu$, with a standard deviation of $sigma$, which must be greater than zero. The $low$ and $high$ parameters give optional artificial bounds on the minimum and maximum output values, respectively.

beta-dist($a, b$) [SAL]
(beta-dist $a b$) [LISP]
Returns a FLONUM value from the Beta distribution. This distribution outputs values between 0 and 1, with outputs more likely to be close to 0 or 1. The parameter $a$ controls the height (probability) of the right side of the distribution (at 1) and $b$ controls the height of the left side (at 0). The distribution is symmetric about 1/2 when $a = b$. 
Figure 14.10: The Beta Distribution, $alpha = .5$, $beta = .25$.

bernoulli-dist($px1$, $x1$, $x2$) [SAL]
(binomial-dist $px1$ $x1$ $x2$) [LISP]
Returns either $x1$ (default value is 1) with probability $px1$ or $x2$ (default value is 0) with probability $1 - px1$. The value of $px1$ should be between 0 and 1. By convention, a result of $x1$ is viewed as a success while $x2$ is viewed as a failure.

binomial-dist($n$, $p$) [SAL]
(binomial-dist $n$ $p$) [LISP]
Returns a FIXNUM value from the binomial distribution, where $n$ is the number of Bernoulli trials run (a FIXNUM) and $p$ is the probability of success in the Bernoulli trial (a FLONUM from 0 to 1). The mean is the product of $n$ and $p$.

group{\textbf{geometric-dist($p$) [SAL]}
\textbf{geometric-dist $p$) [LISP]}
Returns a FIXNUM value from the geometric distribution, which is defined as the number of failures before a success is achieved in a Bernoulli trial with probability of success $p$ (a FLONUM from 0 to 1).

poisson-dist($delta$) [SAL]
(poisson-dist $delta$) [LISP]
Returns a FIXNUM value from the Poisson distribution with a mean of $delta$ (a FIXNUM). The Poisson distribution is often used to generate a sequence of time intervals, resulting in random but often pleasing rhythms.

14.4 Score Generation and Manipulation

A common application of pattern generators is to specify parameters for notes. (It should be understood that "notes" in this context means any Nyquist behavior, whether it represents a conventional note, an abstract sound object, or even some micro-sound event that is just a low-level component of a hierarchical sound organization. Similarly, "score"
Figure 14.11: The Bernoulli Distribution, $p_x I = .75$.

Figure 14.12: The Binomial Distribution, $n = 5, p = .5$. 
Figure 14.13: The Geometric Distribution, $p = .4$.

Figure 14.14: The Poisson Distribution, $\delta = 3$. 
should be taken to mean a specification for a sequence of these "notes." The score-gen macro (defined by loading xm.lsp) establishes a convention for representing scores and for generating them using patterns.

The timed-seq macro, described in Section 7.4, already provides a way to represent a "score" as a list of expressions. The Xmusic representation goes a bit further by specifying that all notes are specified by an alternation of keywords and values, where some keywords have specific meanings and interpretations.

The basic idea of score-gen is you provide a template for notes in a score as a set of keywords and values. For example,

```lisp
set pitch-pattern = make-cycle(list(c4, d4, e4, f4))
score-gen(dur: 0.4, name: quote(my-sound),
          pitch: next(pitch-pattern), score-len: 9)
```

generates a score of 9 notes as follows:

```
((0 0 (SCORE-BEGIN-END 0 3.6))
 (0 0.4 (MY-SOUND :PITCH 60))
 (0.4 0.4 (MY-SOUND :PITCH 62))
 (0.8 0.4 (MY-SOUND :PITCH 64))
 (1.2 0.4 (MY-SOUND :PITCH 65))
 (1.6 0.4 (MY-SOUND :PITCH 60))
 (2 0.4 (MY-SOUND :PITCH 62))
 (2.4 0.4 (MY-SOUND :PITCH 64))
 (2.8 0.4 (MY-SOUND :PITCH 65))
 (3.2 0.4 (MY-SOUND :PITCH 60)))
```

The use of keywords like :PITCH helps to make scores readable and easy to process without specific knowledge of about the functions called in the score. For example, one could write a transpose operation to transform all the :pitch parameters in a score without having to know that pitch is the first parameter of pluck and the second parameter of piano-note. Keyword parameters are also used to give flexibility to note specification with score-gen. Since this approach requires the use of keywords, the next section is a brief explanation of how to define functions that use keyword parameters.

### 14.4.1 Keyword Parameters

Keyword parameters are parameters whose presence is indicated by a special symbol, called a keyword, followed by the actual parameter. Keyword parameters in SAL have default values that are used if no actual parameter is provided by the caller of the function. (See Appendix C to learn about keywords in XLISP.)

To specify that a parameter is a keyword parameter, use a keyword symbol (one that ends in a colon) followed by a default value. For example, here is a function that accepts keyword parameters and invokes the pluck function:

```lisp
define function k-pluck(pitch: 60, dur: 1)
  return pluck(pitch, dur)
```

Now, we can call k-pluck with keyword parameters. The keywords are simply the formal parameter names with a prepended colon character (:pitch and :dur in this example), so a function call would look like:

```lisp
k-pluck(pitch: c3, dur: 3)
```

Usually, it is best to give keyword parameters useful default values. That way, if a parameter such as dur is missing, a reasonable default value (1) can be used automatically. It is never an error to omit a keyword parameter, but the called function can check to see if a keyword parameter was supplied or not. Because of default values, we can call
In XLISP, there is additional syntax to specify an alternate symbol to be used as the keyword and to allow the called function to determine whether or not a keyword parameter was supplied, but these features are little-used. See the XLISP manual for details.

**14.4.2 Using score-gen**

The `score-gen` macro computes a score based on keyword parameters. Some keywords have a special meaning, while others are not interpreted but merely placed in the score. The resulting score can be synthesized using `timed-seq` (see Section 7.4).

The form of a call to `score-gen` is simply:

```
(score-gen :k1 e1 :k2 e2 ...)
```

where the `k`'s are keywords and the `e`'s are expressions. A score is generated by evaluating the expressions once for each note and constructing a list of keyword-value pairs. A number of keywords have special interpretations.

The rules for interpreting these parameters will be explained through a set of "How do I..." questions below.

**How many notes will be generated?** The keyword parameter `score-len:` specifies an upper bound on the number of notes. (Note: in LISP syntax, keywords are always preceded by colons, so you would write `:score-len` instead.) The keyword `score-dur:` specifies an upper bound on the starting time of the last note in the score. (To be more precise, the `score-dur:` bound is reached when the default starting time of the next note is greater than or equal to the `score-dur:` value. This definition is necessary because note times are not strictly increasing.) When either bound is reached, score generation ends. At least one of these two parameters must be specified or an error is raised. These keyword parameters are evaluated just once and are not copied into the parameter lists of generated notes.

**What is the duration of generated notes?** The keyword `dur:` defaults to 1 and specifies the nominal duration in seconds. Since the generated note list is compatible with `timed-seq`, the starting time and duration (to be precise, the stretch factor) are not passed as parameters to the notes. Instead, they control the Nyquist environment in which the note will be evaluated.

**What is the start time of a note?** The default start time of the first note is zero. Given a note, the default start time of the next note is the start time plus the inter-onset time, which is given by the `ioi:` parameter. If no `ioi:` parameter is specified, the inter-onset time defaults to the duration, given by `dur:`. In all cases, the default start time of a note can be overridden by the keyword parameter `time:`.

**When does the score begin and end?** The behavior `SCORE-BEGIN-END` contains the beginning and ending of the score (these are used for score manipulations, e.g. when scores are merged, their begin times can be aligned.) When `timed-seq` is used to synthesize a score, the `SCORE-BEGIN-END` marker is not evaluated. The `score-gen` macro inserts a "note" of the form `(0 0 (SCORE-BEGIN-END begin-time end-time))` at the time given by the `begin:` keyword, with `begin-time` and `end-time` determined by the `begin:` and `end:` keyword parameters, respectively. If the `begin:` keyword is not provided, the score begins at zero. If the `end:` keyword is not provided, the score ends at the default start time of what would be the next note after the last note in the score (as described in the previous paragraph). Note: if `time:` is used to compute note starting times, and these times are not increasing, it is strongly advised to use `end:` to specify an end time for the score, because the default end time may be anywhere in the middle of the generated sequence.
What function is called to synthesize the note? The name: parameter names the function. Like other parameters, the value can be any expression, including something like next(fn-name-pattern), allowing function names to be recomputed for each note. The default value is note.

Can I make parameters depend upon the starting time or the duration of the note? Parameter expressions can use the variable sg:start to access the start time of the note, sg:ioi to access the inter-onset time, and sg:dur to access the duration (stretch factor) of the note. Also, sg:count counts how many notes have been computed so far, starting at 0. The order of computation is: sg:start first, then sg:ioi and sg:dur, so for example, an expression to compute sg:dur can depend on sg:ioi.

Can parameters depend on each other? The keyword pre: introduces an expression that is evaluated before each note, and post: provides an expression to be evaluated after each note. The pre: expression can assign one or more global variables which are then used in one or more expressions for parameters.

How do I debug score-gen expressions? You can set the trace: parameter to true (t) to enable a print statement for each generated note.

How can I save scores generated by score-gen that I like? If the keyword parameter save: is set to a symbol, the global variable named by the symbol is set to the value of the generated sequence. Of course, the value returned by score-gen is just an ordinary list that can be saved like any other value.

In summary, the following keywords have special interpretations in score-gen: begin:, end:, time:, dur:, name:, ioi:, trace:, save:, score-len:, score-dur:, pre:, post:. All other keyword parameters are expressions that are evaluated once for each note and become the parameters of the notes.

14.4.3 Score Manipulation

Nyquist encourages the representation of music as executable programs, or behaviors, and there are various ways to modify behaviors, including time stretching, transposition, etc. An alternative to composing executable programs is to manipulate scores as editable data. Each approach has its strengths and weaknesses. This section describes functions intended to manipulate Xmusic scores as generated by, or at least in the form generated by, score-gen. Recall that this means scores are lists of events (e.g. notes), where events are three-element lists of the form (time duration expression, and where expression is a standard lisp function call where all parameters are keyword parameters. In addition, the first "note" may be the special SCORE-BEGIN-END expression. If this is missing, the score begins at zero and ends at the end of the last note.

For convenience, a set of functions is offered to access properties of events (or notes) in scores. Although lisp functions such as car, cadr, and caddr can be used, code is more readable when more mnemonic functions are used to access events.

```
(event-time event) [SAL]
(event-time event) [LISP]
Retrieve the time field from an event.

(event-set-time event time) [SAL]
(event-set-time event time) [LISP]
Construct a new event where the time of event is replaced by time.

(event-dur event) [SAL]
(event-dur event) [LISP]
Retrieve the duration (i.e. the stretch factor) field from an event.
```
event-set-dur(event, dur) [SAL]
(event-set-dur event dur) [LISP]
Construct a new event where the duration (or stretch factor) of event is replaced by dur.

event-expression(event) [SAL]
(event-expression event) [LISP]
Retrieve the expression field from an event.

event-set-expression(event, dur) [SAL]
(event-set-expression event dur) [LISP]
Construct a new event where the expression of event is replaced by expression.

event-end(event) [SAL]
(event-end event) [LISP]
Retrieve the end time of event, its time plus its duration.

expr-has-attr(expression, attribute) [SAL]
(expr-has-attr expression attribute) [LISP]
Test whether a score event expression has the given attribute.

expr-get-attr(expression, attribute [, default]) [SAL]
(expr-get-attr expression attribute [default]) [LISP]
Get the value of the given attribute from a score event expression. If attribute is not present, return default if specified, and otherwise nil.

expr-set-attr(expr, attribute, value) [SAL]
(expr-set-attr expr attribute value) [LISP]
Construct a new expression identical to expr except that the attribute has value.

event-has-attr(event, attribute) [SAL]
(event-has-attr event attribute) [LISP]
Test whether a given score event’s expression has the given attribute.

event-get-attr(event, attribute, [default]) [SAL]
(event-get-attr event attribute [default]) [LISP]
Get the value of the given attribute from a score event’s expression. If attribute is not present, return default if specified, and otherwise nil.

event-set-attr(event, attribute, value) [SAL]
(event-set-attr event attribute value) [LISP]
Construct a new event identical to event except that the attribute has value.

Functions are provided to shift the starting times of notes, stretch times and durations, stretch only durations, add an offset to a keyword parameter, scale a keyword parameter, and other manipulations. Functions are also provided to extract ranges of notes, notes that match criteria, and to combine scores. Most of these functions (listed below in detail) share a set of keyword parameters that optionally limit the range over which the transformation operates. The from-index: and to-index: parameters specify the index of the first note and the index of the last note to be changed. If these numbers are negative, they are offsets from the end of the score, e.g. -1 denotes the last note of the score. The from-time: and to-time: indicate a range of starting times of notes that will be affected by the manipulation. Only notes whose time is greater than or equal to the from-time and strictly less than the to-time are modified. If both index and time ranges are specified, only notes that satisfy both constraints are selected. (Note: in LISP syntax, colons precede the keyword, so use :from-index, :to-index, :from-time, and :to-time.)
score-sorted(score) [SAL]
(score-sorted score) [LISP]
    Test if score is sorted.

(score-sort(score [, copy-flag])) [SAL]
(score-sort score [copy-flag]) [LISP]
    Sort the notes in a score into start-time order. If copy-flag is nil, this is a destructive operation which should only be performed if the top-level score list is a fresh copy that is not shared by any other variables. (The copy-flag is intended for internal system use only.) For the following operations, it is assumed that scores are sorted, and all operations return a sorted score.

(score-shift(score, offset, from-index: i, to-index: j, from-time: x, to-time: y)) [SAL]
(score-shift score offset :from-index i :to-index j :from-time x :to-time y) [LISP]
    Add a constant offset to the starting time of a set of notes in score. By default, all notes are modified, but the range of notes can be limited with the keyword parameters. The begin time of the score is not changed, but the end time is increased by offset. The original score is not modified, and a new score is returned.

(score-stretch(score, factor, dur: dur-flag, time: time-flag, from-index: i, to-index: j, from-time: x, to-time: y)) [SAL]
(score-stretch score factor :dur dur-flag :time time-flag :from-index i :to-index j :from-time x :to-time y) [LISP]
    Stretch note times and durations by factor. The default dur-flag is non-null, but if dur-flag is null, the original durations are retained and only times are stretched. Similarly, the default time-flag is non-null, but if time-flag is null, the original times are retained and only durations are stretched. If both dur-flag and time-flag are null, the score is not changed. If a range of notes is specified, times are scaled within that range, and notes after the range are shifted so that the stretched region does not create a "hole" or overlap with notes that follow. If the range begins or ends with a time (via from-time: and to-time:), time stretching takes place over the indicated time interval independent of whether any notes are present or where they start. In other words, the "rests" are stretched along with the notes. The original score is not modified, and a new score is returned.

(score-transpose(score, keyword, amount, from-index: i, to-index: j, from-time: x, to-time: y)) [SAL]
(score-transpose score keyword amount :from-index i :to-index j :from-time x :to-time y) [LISP]
    For each note in the score and in any indicated range, if there is a keyword parameter matching keyword and the parameter value is a number, increment the parameter value by amount. For example, to transpose up by a whole step, write (score-transpose 2 :pitch score). The original score is not modified, and a new score is returned.

(score-scale(score, keyword, amount, from-index: i, to-index: j, from-time: x, to-time: y)) [SAL]
(score-scale score keyword amount :from-index i :to-index j :from-time x :to-time y) [LISP]
    For each note in the score and in any indicated range, if there is a keyword parameter matching keyword and the parameter value is a number, multiply the parameter value by amount. The original score is not modified, and a new score is returned.

(score-sustain(score, factor, from-index: i, to-index: j, from-time: x, to-time: y)) [SAL]
(score-sustain score factor :from-index i :to-index j :from-time x :to-time y) [LISP]
    For each note in the score and in any indicated range, multiply the duration (stretch factor) by amount. This can be used to make notes sound more legato or staccato, and does not change their starting times. The original score is not modified, and a new score is returned.
score-voice(score, replacement-list, from-index: i, to-index: j, from-time: x, to-time: y) [SAL]
(score-voice score replacement-list :from-index i :to-index j :from-time x :to-time y) [LISP]
For each note in the score and in any indicated range, replace the behavior (function) name using replacement-list, which has the format: ((old1 new1) (old2 new2) ...), where oldi indicates a current behavior name and newi is the replacement. If oldi is *, it matches anything. For example, to replace my-note-1 by trombone and my-note-2 by horn, use score-voice(score, {{my-note-1 trombone} {my-note-2 horn}}). To replace all instruments with piano, use score-voice(score, {{* piano}}). The original score is not modified, and a new score is returned.

score-merge(score1, score2, ...) [SAL]
(score-merge score1 score2 ...) [LISP]
Create a new score containing all the notes of the parameters, which are all scores. The resulting notes retain their original times and durations. The merged score begin time is the minimum of the begin times of the parameters and the merged score end time is the maximum of the end times of the parameters. The original scores are not modified, and a new score is returned.

score-append(score1, score2, ...) [SAL]
(score-append score1 score2 ...) [LISP]
Create a new score containing all the notes of the parameters, which are all scores. The begin time of the first score is unaltered. The begin time of each other score is aligned to the end time of the previous score; thus, scores are "spliced" in sequence. The original scores are not modified, and a new score is returned.

score-select(score, predicate, from-index: i, to-index: j, from-time: x, to-time: y, reject: flag) [SAL]
(score-select score predicate :from-index i :to-index j :from-time x :to-time y :reject flag) [LISP]
Select (or reject) notes to form a new score. Notes are selected if they fall into the given ranges of index and time and they satisfy predicate, a function of three parameters that is applied to the start time, duration, and the expression of the note. Alternatively, predicate may be t, indicating that all notes in range are to be selected. The selected notes along with the existing score begin and end markers, are combined to form a new score. Alternatively, if the reject: parameter is non-null, the notes not selected form the new score (in other words the selected notes are rejected or removed to form the new score). The original score is not modified, and a new score is returned.

score-set-begin(score, time) [SAL]
(score-set-begin score time) [LISP]
The begin time from the score’s SCORE-BEGIN-END marker is set to time. The original score is not modified, and a new score is returned.

score-get-begin(score) [SAL]
(score-get-begin score) [LISP]
Return the begin time of the score.

score-set-end(score, time) [SAL]
(score-set-end score time) [LISP]
The end time from the score’s SCORE-BEGIN-END marker is set to time. The original score is not modified, and a new score is returned.
 Nyquist Reference Manual   Chapter 14. Xmusic and Algorithmic Composition

score-get-end(score) [SAL]
(score-get-end score) [LISP]
   Return the end time of the score.

score-must-have-begin-end(score) [SAL]
(score-must-have-begin-end score) [LISP]
   If score does not have a begin and end time, construct a score with a SCORE-BEGIN-END expression and return it. If score already has a begin and end time, just return the score. The original score is not modified.

score-filter-length(score, cutoff) [SAL]
(score-filter-length score cutoff) [LISP]
   Remove notes that extend beyond the cutoff time. This is similar to score-select, but the here, events are removed when their nominal ending time (start time plus duration) exceeds the cutoff, whereas the to-time: parameter is compared to the note’s start time. The original score is not modified, and a new score is returned.

score-repeat(score, n) [SAL]
(score-repeat score n) [LISP]
   Make a sequence of n copies of score. Each copy is shifted to that it’s begin time aligns with the end time of the previous copy, as in score-append. The original score is not modified, and a new score is returned.

score-stretch-to-length(score, length) [SAL]
(score-stretch-to-length score length) [LISP]
   Stretch the score so that the end time of the score is the score’s begin time plus length. The original score is not modified, and a new score is returned.

score-filter-overlap(score) [SAL]
(score-filter-overlap score) [LISP]
   Remove overlapping notes (based on the note start time and duration), giving priority to the positional order within the note list (which is also time order). The original score is not modified, and a new score is returned.

score-print(score) [SAL]
(score-print score) [LISP]
   Print a score with one note per line. Returns nil.

score-play(score) [SAL]
(score-play score) [LISP]
   Play score using timed-seq to convert the score to a sound, and play to play the sound.

score-adjacent-events(score, function, from-index: i, to-index: j, from-time: x, to-time: y) [SAL]
(score-adjacent-events score function :from-index i :to-index j :from-time x :to-time y) [LISP]
   Call (function A B C), where A, B, and C are consecutive notes in the score. The result replaces B. If the result is nil, B is deleted, and the next call will be (function A C D), etc. The first call is to (function nil A B) and the last is to (function Y Z nil). If there is just one note in the score, (function nil A nil) is called. Function calls are not made if the note is outside of the indicated range. This function allows notes and their parameters to be adjusted according to their immediate context. The original score is not modified, and a new score is returned.

score-apply(score, function, from-index: i, to-index: j, from-time: x, to-time: y) [SAL]
(score-apply score function :from-index i :to-index j :from-time x :to-time y) [LISP]
   Replace each note in the score with the result of (function time dur expression) (in Lisp) or function(time, dur, expression) (in SAL), where time, dur, and expression are the time, duration, and expression of the note. If a range is indicated, only notes in the range are replaced. The original score is not modified, and a new score is returned.
score-indexof(score, function, from-index: i, to-index: j, from-time: x, to-time: y) [SAL]
(score-indexof score function :from-index i :to-index j :from-time x :to-time y) [LISP]
Return the index (position) of the first score event (in range) for which applying function using (function time dur expression) returns true.

score-last-indexof(score, function, from-index: i, to-index: j, from-time: x, to-time: y) [SAL]
(score-last-indexof score function :from-index i :to-index j :from-time x :to-time y) [LISP]
Return the index (position) of the last score event (in range) for which applying function using (function time dur expression) returns true.

score-randomize-start(score, amt, from-index: i, to-index: j, from-time: x, to-time: y) [SAL]
(score-randomize-start score amt :from-index i :to-index j :from-time x :to-time y) [LISP]
Alter the start times of notes by a random amount up to plus or minus amt. The original score is not modified, and a new score is returned.

14.4.4 Xmusic and Standard MIDI Files

Nyquist has a general facility to read and write MIDI files. You can even translate to and from a text representation, as described in Chapter 11. It is also useful sometimes to read notes from Standard MIDI Files into Xmusic scores and vice versa. At present, Xmusic only translates notes, ignoring the various controls, program changes, pitch bends, and other messages.

MIDI notes are translated to Xmusic score events as follows:

\[(\text{time dur} \ (\text{NOTE :chan channel :pitch keynum :vel velocity}))\]

where channel, keynum, and velocity come directly from the MIDI message (channels are numbered starting from zero).

Note also that note-off messages are implied by the stretch factor dur which is duration in seconds.

score-read-smf(filename) [SAL]
(score-read-smf filename) [LISP]
Read a standard MIDI file from filename. Return an Xmusic score, or nil if the file could not be opened. The start time is zero, and the end time is the maximum end time of all notes. A very limited interface is offered to extract MIDI program numbers from the file: The global variable *rslt* is set to a list of MIDI program numbers for each channel. E.g. if *rslt* is (0 20 77), then program for channel 0 is 0, for channel 1 is 20, and for channel 2 is 77. Program changes were not found on other channels. The default program number is 0, so in this example, it is not known whether the program 0 on channel 0 is the result of a real MIDI program change command or just a default value. If more than one program change exists on a channel, the last program number is recorded and returned, so this information will only be completely correct when the MIDI file sends single program change per channel before any notes are played. This, however, is a fairly common practice. Note that the list returned as *rslt* can be passed to score-write-smf, described below.
score-read(filename) [SAL]
(score-read filename) [LISP]
Read an Adagio file from filename. Return an Xmusic score, or nil if the file could not be opened. See Chapter 11 for details on Adagio, a text-based score language. See score-read-smf for details on handling program changes.

score-write-smf(score, filename, [programs as-adagio]) [SAL]
(score-write-smf score filename [programs as-adagio]) [LISP]
Write a standard MIDI file to filename with notes in score. In this function, every event in the score with a pitch: attribute, regardless of the “instrument” (or function name), generates a MIDI note, using the chan: attribute for the channel (default 0) and the vol: attribute for velocity (default 100). There is no facility (in the current implementation) to issue control changes, but to allow different instruments, MIDI programs may be set in two ways. The simplest is to associate programs with channels using the optional programs parameter, which is simply a list of up to 16 MIDI program numbers. Corresponding program change commands are added to the beginning of the MIDI file. If programs has less than 16 elements, program change commands are only sent on the first n channels. The second way to issue MIDI program changes is to add a program: keyword parameter to a note in the score. Typically, the note will have a pitch: of nil so that no actual MIDI note-on message is generated. If program changes and notes have the same starting times, their relative playback order is undefined, and the note may be cut off by an immediately following program change. Therefore, program changes should occur slightly, e.g. 1 ms, before any notes. Program numbers and channels are numbered starting at zero, matching the internal MIDI representation. This may be one less than displayed on MIDI hardware, sequencers, etc. The as-adagio optional parameter should normally be omitted. If non-nil, the file is written in Adagio format, but if you want to do that, call score-write instead.

score-write(score, filename, [programs]) [SAL]
(score-write score filename [programs]) [LISP]
Write an Adagio format file to filename with notes in score. See Chapter 11 for details on Adagio, a text-based score language. See score-write-smf for details on MIDI program changes.

14.4.5 Workspaces

When working with scores, you may find it necessary to save them in files between work sessions. This is not an issue with functions because they are normally edited in files and loaded from them. In contrast, scores are created as Lisp data, and unless you take care to save them, they will be destroyed when you exit the Nyquist program.

A simple mechanism called a workspace has been created to manage scores (and any other Lisp data, for that matter). A workspace is just a set of Lisp global variables. These variables are stored in the file workspace.lsp. For simplicity, there is only one workspace, and no backups or versions are maintained, but the user is free to make backups and copies of workspace.lsp. To help remember what each variable is for, you can also associate and retrieve a text string with each variable. The following functions manage workspaces.

In addition, when a workspace is loaded, you can request that functions be called. For example, the workspace might store descriptions of a graphical interface. When the workspace is loaded, a function might run to convert saved data into a graphical interface. (This is how sliders are saved by the IDE.)

add-to-workspace(symbol) [SAL]
(add-to-workspace symbol) [LISP]
Adds a global variable to the workspace. The symbol should be a (quoted) symbol.
Chapter 14. Xmusic and Algorithmic Composition

save-workspace() [SAL]
(save-workspace) [LISP]
   All global variables in the workspace are saved to workspace.lsp (in the current directory), overwriting the previous file.

describe(symbol [, description]) [SAL]
(describe symbol [description]) [LISP]
   If description, a text string, is present, associate description with the variable named by the symbol. If symbol is not already in the workspace, it is added. If description is omitted, the function returns the current description (from a previous call) for symbol.

add-action-to-workspace(symbol) [SAL]
(add-action-to-workspace symbol) [LISP]
   Requests that the function named by symbol be called when the workspace is loaded (if the function is defined).

To restore a workspace, use the command load "workspace". This restores the values of the workspace variables to the values they had when save-workspace was last called. It also restores the documentation strings, if set, by describe. If you load two or more workspace.lsp files, the variables will be merged into a single workspace. The current set of workspace variables are saved in the list *workspace*. To clear the workspace, set *workspace* to nil. This does not delete any variables, but means that no variables will be saved by save-workspace until variables are added again.

Functions to be called are saved in the list *workspace-actions*. To clear the functions, set *workspace-actions* to nil. Restore functions to the list with add-action-to-workspace.

14.4.6 Utility Functions

This chapter concludes with details of various utility functions for score manipulation.

patternp(expression) [SAL]
(patternp expression) [LISP]
   Test if expression is an Xmusic pattern.

params-transpose(params, keyword, amount) [SAL]
(params-transpose params keyword amount) [LISP]
   Add a transposition amount to a score event parameter. The params parameter is a list of keyword/value pairs (not preceded by a function name). The keyword is the keyword of the value to be altered, and amount is a number to be added to the value. If no matching keyword is present in params, then params is returned. Otherwise, a new parameter list is constructed and returned. The original params is not changed.

params-scale(params, keyword, amount) [SAL]
(params-scale params keyword amount) [LISP]
   Scale a score event parameter by some factor. This is like params-transpose, only using multiplication. The params list is a list of keyword/value pairs, keyword is the parameter keyword, and amount is the scale factor.

interpolate(x, x1, y1, x2, y2) [SAL]
(interpolate x x1 y1 x2 y2) [LISP]
   Linearly interpolate (or extrapolate) between points (x1, y1) and (x2, y2) to compute the y value corresponding to x.
intersection(a, b) [SAL]
(intersection a b) [LISP]
Compute the set intersection of lists a and b.

union(a, b) [SAL]
(union a b) [LISP]
Compute the set union of lists a and b.

set-difference(a, b) [SAL]
(set-difference a b) [LISP]
Compute the set of all elements that are in a but not in b.

subsetp(a, b) [SAL]
(subsetp a b) [LISP]
Returns true iff a is a subset of b, that is, each element of a is a member of b.
15 Nyquist Libraries

Nyquist is always growing with new functions. Functions that are most fundamental are added to the core language. These functions are automatically loaded when you start Nyquist, and they are documented in the preceding chapters. Other functions seem less central and are implemented as lisp files that you can load. These are called library functions, and they are described here.

To use a library function, you must first load the library, e.g. `(load "pianosyn")` loads the piano synthesis library. The libraries are all located in the `lib` directory, and you should therefore include this directory on your `XLISPPATH` variable. (See Section 1.1.) Each library is documented in one of the following sections. When you load the library described by the section, all functions documented in that section become available.

15.1 Statistics

The file `statistics.lsp` defines a class and functions to compute simple statistics, histograms, correlation, and some other tests. See the source code for complete details.

15.2 Plots

The Nyquist IDE has a simple facility to plot signals. For more advanced plotting, you can use `gnuplot.sal` to generate plots for `gnuplot`, a separate, but free program. See the source for details.

15.3 Labeling Audio Events, Marking Audio Times, Displaying Marked Audio Times

The `labels.sal` program can convert lists to label files and label files to lists. Label files can be loaded along with audio in Audacity to show metadata. See the source for details.
15.4 Linear Regression

See regression.sal for simple linear regression functions.

15.5 Vector Math, Linear Algebra

See vectors.lsp for a simple implementation of vector arithmetic and other vector functions.

15.6 Piano Synthesizer

The piano synthesizer (library name is pianosyn.lsp) generates realistic piano tones using a multiple wavetable implementation by Zheng (Geoffrey) Hua and Jim Beauchamp, University of Illinois. Please see the notice about acknowledgements that prints when you load the file. Further informations and example code can be found in demos/piano.htm. There are several useful functions in this library. These functions auto-load the pianosyn.lsp library if it is not already loaded:

\[
piano\text{-}note(duration, step, dynamic)\quad [\text{SAL}]
\]

\[
(\text{piano\text{-}note } duration \text{ step } dynamic)\quad [\text{LISP}]
\]

Synthesizes a piano tone. Duration is the duration to the point of key release, after which there is a rapid decay. Step is the pitch in half steps, and dynamic is approximately equivalent to a MIDI key velocity parameter. Use a value near 100 for a loud sound and near 10 for a soft sound.

\[
piano\text{-}note\text{-}2(step, dynamic)\quad [\text{SAL}]
\]

\[
(\text{piano\text{-}note\text{-}2 } step \text{ dynamic})\quad [\text{LISP}]
\]

Similar to piano\text{-}note except the duration is nominally 1.0.

\[
piano\text{-}midi(midi\text{-}file\text{-}name)\quad [\text{SAL}]
\]

\[
(\text{piano\text{-}midi } midi\text{-}file\text{-}name)\quad [\text{LISP}]
\]

Use the piano synthesizer to play a MIDI file. The file name (a string) is given by midi\text{-}file\text{-}name.

\[
piano\text{-}midi2file(midi\text{-}file\text{-}name, sound\text{-}file\text{-}name)\quad [\text{SAL}]
\]

\[
(\text{piano\text{-}midi2file } midi\text{-}file\text{-}name \text{ sound\text{-}file\text{-}name})\quad [\text{LISP}]
\]

Use the piano synthesizer to play a MIDI file. The MIDI file is given by midi\text{-}file\text{-}name and the (monophonic) result is written to the file named sound\text{-}file\text{-}name.

15.7 Dynamics Compression

These functions implement a compressor originally intended for noisy speech audio, but usable in a variety of situations. There are actually two compressors that can be used in series. The first, compress, is a fairly standard one: it detects signal level with an RMS detector and uses table-lookup to determine how much gain to place on the original signal at that point. One bit of cleverness here is that the RMS envelope is "followed" or enveloped using snd\text{-}follow, which does look-ahead to anticipate peaks before they happen.
The other interesting feature is \texttt{compress-map}, which builds a map in terms of compression and expansion. For speech, the recommended procedure is to figure out the noise floor on the signal you are compressing (for example, look at the signal where the speaker is not talking). Use a compression map that leaves the noise alone and boosts signals that are well above the noise floor. Alas, the \texttt{compress-map} function is not written in these terms, so some head-scratching is involved, but the results are quite good.

The second compressor is called \texttt{agc}, and it implements automatic gain control that keeps peaks at or below 1.0. By combining \texttt{compress} and \texttt{agc}, you can process poorly recorded speech for playback on low-quality speakers in noisy environments. The \texttt{compress} function modulates the short-term gain to to minimize the total dynamic range, keeping the speech at a generally loud level, and the \texttt{agc} function rides the long-term gain to set the overall level without clipping.

\begin{verbatim}
compress-map(compress-ratio, compress-threshold, expand-ratio, expand-threshold, limit: limit, transition: transition, verbose: verbose) [SAL]
\end{verbatim}

\begin{verbatim}
(compress-map compress-ratio compress-threshold expand-ratio expand-threshold :limit limit :transition transition :verbose verbose) [LISP]
\end{verbatim}

Construct a map for the compress function. The map consists of two parts: a compression part and an expansion part. The intended use is to compress everything above compress-threshold by compress-ratio, and to downward expand everything below expand-ratio by expand-ratio. Thresholds are in dB and ratios are dB-per-dB. 0dB corresponds to a peak amplitude of 1.0 or rms amplitude of 0.7 If the input goes above 0dB, the output can optionally be limited by setting limit: (a keyword parameter) to \texttt{T}. This effectively changes the compression ratio to infinity at 0dB. If limit: is \texttt{nil} (the default), then the compression-ratio continues to apply above 0dB.

Another keyword parameter, transition:, sets the amount below the thresholds (in dB) that a smooth transition starts. The default is 0, meaning that there is no smooth transition. The smooth transition is a 2nd-order polynomial that matches the slopes of the straight-line compression curve and interpolates between them.

If verbose is true (this is the default), the map is printed, showing, for each dB value below zero of this input, what is the gain (in dB) indicated by the output. Only regions where the map is changing are printed because at lower values, the dB gain is constant.

It is assumed that expand-threshold <= compress-threshold <= 0 The gain is unity at 0dB so if compression-ratio > 1, then gain will be greater than unity below 0dB.

The result returned by this function is a sound for use in the shape function. The sound maps input dB to gain. Time 1.0 corresponds to 0dB, time 0.0 corresponds to -100 dB, and time 2.0 corresponds to +100dB, so this is a 100hz "sample rate" sound. The sound gives gain in dB.

\begin{verbatim}
db-average(input) [SAL]
db-average input [LISP]
\end{verbatim}

Compute the average amplitude of \texttt{input} in dB.

\begin{verbatim}
compress(input, map, rise-time, fall-time [ , lookahead]) [SAL]
\end{verbatim}

\begin{verbatim}
(compress input map rise-time fall-time [lookahead]) [LISP]
\end{verbatim}

Compress \texttt{input} using \texttt{map}, a compression curve probably generated by \texttt{compress-map} (see above). Adjustments in gain have the given \texttt{rise-time} and \texttt{fall-time}. Lookahead tells how far ahead to look at the signal, and is \texttt{rise-time} by default.

\begin{verbatim}
agc(input, range, rise-time, fall-time [ , lookahead]) [SAL]
\end{verbatim}

\begin{verbatim}
(agc input range rise-time fall-time [lookahead]) [LISP]
\end{verbatim}

An automatic gain control applied to \texttt{input}. The maximum gain in dB is \texttt{range}. Peaks are attenuated to 1.0, and gain is controlled with the given \texttt{rise-time} and \texttt{fall-time}. The look-ahead time default is \texttt{rise-time}.
15.8  Clipping Softener

This library, in `soften.lsp`, was written to improve the quality of poorly recorded speech. In recordings of speech, extreme clipping generates harsh high frequency noise. This can sound particulary bad on small speakers that will emphasize high frequencies. This problem can be ameliorated by low-pass filtering regions where clipping occurs. The effect is to dull the harsh clipping. Intelligibility is not affected by much, and the result can be much more pleasant on the ears. Clipping is detected simply by looking for large signal values. Assuming 8-bit recording, this level is set to 126/127.

The function works by cross-fading between the normal signal and a filtered signal as opposed to changing filter coefficients.

```
soften-clipping(snd, cutoff) [SAL]
(soften-clipping snd cutoff) [LISP]
```

Filter the loud regions of a signal where clipping is likely to have generated additional high frequencies. The input signal is `snd` and `cutoff` is the filter cutoff frequency (4 kHz is recommended for speech).

15.9  Graphical Equalizer

There’s nothing really “graphical” about this library (`grapheq.lsp`), but this is a common term for multi-band equalizers. This implementation uses Nyquist’s `eq-band` function to split the incoming signal into different frequency bands. Bands are spaced geometrically, e.g. each band could be one octave, meaning that each successive band has twice the bandwidth. An interesting possibility is using computed control functions to make the equalization change over time.

```
nband-range(input, gains, lowf, highf) [SAL]
(nband-range input gains lowf highf) [LISP]
```

A graphical equalizer applied to `input` (a `SOUND`). The gain controls and number of bands is given by `gains`, an `ARRAY` of `SOUNDS` (in other words, a Nyquist multichannel `SOUND`). Any sound in the array may be replaced by a `FLONUM`. The bands are geometrically equally spaced from the lowest frequency `lowf` to the highest frequency `highf` (both are `FLONUMs`).

```
nband(input, gains) [SAL]
(nband input gains) [LISP]
```

A graphical equalizer, identical to `nband-range` with a range of 20 to 20,000 Hz.

15.10  Sound Reversal

The `reverse.lsp` library implements functions to play sounds in reverse.
s-reverse(snd) [SAL]
(s-reverse snd) [LISP]
Reverses snd (a SOUND). Sound must be shorter than *max-reverse-samples*, which is currently initialized to 25 million samples. Reversal allocates about 4 bytes per sample. This function uses XLISP in the inner sample loop, so do not be surprised if it calls the garbage collector a lot and runs slowly. The result starts at the starting time given by the current environment (not necessarily the starting time of snd). If snd has multiple channels, a multiple channel, reversed sound is returned.

s-read-reverse(filename, time-offset: offset, srate: sr, dur: dur,
   nchans: chans, format: format, mode: mode, bits: n, swap: flag) [SAL]
(s-read-reverse filename :time-offset offset :srate sr :dur dur
   :nchans chans :format format :mode mode :bits n :swap flag) [LISP]
This function is identical to s-read (see Section 7.5), except it reads the indicated samples in reverse. Like s-reverse (see above), it uses XLISP in the inner loop, so it is slow. Unlike s-reverse, s-read-reverse uses a fixed amount of memory that is independent of how many samples are computed. Multiple channels are handled.

15.11 Time Delay Functions

The time-delay-fns.lsp library implements chorus, phaser, and flange effects.

phaser(snd) [SAL]
(phaser snd) [LISP]
A phaser effect applied to snd (a SOUND). There are no parameters, but feel free to modify the source code of this one-liner.

flange(snd) [SAL]
(flange snd) [LISP]
A flange effect applied to snd. To vary the rate and other parameters, see the source code.

stereo-chorus(snd, delay: delay, depth: depth, rate1: rate1, rate2: rate2
   saturation: saturation) [SAL]
(stereo-chorus snd :delay delay :depth depth :rate1 rate1 :rate2 rate2
   :saturation saturation) [LISP]
A chorus effect applied to snd, a SOUND (monophonic). The output is a stereo sound with out-of-phase chorus effects applied separately for the left and right channels. See the chorus function below for a description of the optional parameters. The rate1 and rate2 parameters are rate parameters for the left and right channels.

chorus(snd, delay: delay, depth: depth, rate: rate, saturation: saturation, phase: phase) [SAL]
(chorus snd :delay delay :depth depth :rate rate :saturation saturation :phase phase) [LISP]
A chorus effect applied to snd. All parameters may be arrays as usual. The chorus is implemented as a variable delay modulated by a sinusoid shifted by phase degrees (a FLONUM) oscillating at rate Hz (a FLONUM). The sinusoid is scaled by depth (a FLONUM. The delayed signal is mixed with the original, and saturation (a FLONUM) gives the fraction of the delayed signal (from 0 to 1) in the mix. Default values are delay 0.03, depth 0.003, rate 0.3, saturation 1.0, and phase 0.0 (degrees).
15.12 Multiple Band Effects

The bandfx.lsp library implements several effects based on multiple frequency bands. The idea is to separate a signal into different frequency bands, apply a slightly different effect to each band, and sum the effected bands back together to form the result. This file includes its own set of examples. After loading the file, try f2(), f3(), f4(), and f5() to hear them. Further discussion and examples can be found in demos/bandfx.htm.

There is much room for expansion and experimentation with this library. Other effects might include distortion in certain bands (for example, there are commercial effects that add distortion to low frequencies to enhance the sound of the bass), separating bands into different channels for stereo or multi-channel effects, adding frequency-dependent reverb, and performing dynamic compression, limiting, or noise gate functions on each band. There are also opportunities for cross-synthesis: using the content of bands extracted from one signal to modify the bands of another. The simplest of these would be to apply amplitude envelopes of one sound to another. Please contact us (dannenberg@cs.cmu.edu) if you are interested in working on this library.

apply-banded-delay(s, lowp, highp, num-bands, loud, highd, fb, wet) [SAL]
(apply-banded-delay s lowp highp num-bands loud highd fb wet) [LISP]
Separates input SOUND s into FIXNUM num-bands bands from a low frequency of lowp to a high frequency of highp (these are FLONUMS that specify steps, not Hz), and applies a delay to each band. The delay for the lowest band is given by the FLONUM lowd (in seconds) and the delay for the highest band is given by the FLONUM highd. The delays for other bands are linearly interpolated between these values. Each delay has feedback gain controlled by FLONUM fb. The delayed bands are scaled by FLONUM wet, and the original sound is scaled by 1 - wet. All are summed to form the result, a SOUND.

apply-banded-bass-boost(s, lowp, highp, num-bands, num-boost, gain) [SAL]
(apply-banded-bass-boost s lowp highp num-bands num-boost gain) [LISP]
Applies a boost to low frequencies. Separates input SOUND s into FIXNUM num-bands bands from a low frequency of lowp to a high frequency of highp (these are FLONUMS that specify steps, not Hz), and scales the lowest num-boost (a FIXNUM) bands by gain, a FLONUM. The bands are summed to form the result, a SOUND.

apply-banded-treble-boost(s, lowp, highp, num-bands, num-boost, gain) [SAL]
(apply-banded-treble-boost s lowp highp num-bands num-boost gain) [LISP]
Applies a boost to high frequencies. Separates input SOUND s into FIXNUM num-bands bands from a low frequency of lowp to a high frequency of highp (these are FLONUMS that specify steps, not Hz), and scales the highest num-boost (a FIXNUM) bands by gain, a FLONUM. The bands are summed to form the result, a SOUND.

15.13 Granular Synthesis

Some granular synthesis functions are implemented in the gran.lsp library file. There are many variations and control schemes one could adopt for granular synthesis, so it is impossible to create a single universal granular synthesis function. One of the advantages of Nyquist is the integration of control and synthesis functions, and users are encouraged to build their own granular synthesis functions incorporating their own control schemes. The gran.lsp file includes many comments and is intended to be a useful starting point. Another possibility is to construct a score with an event for each grain. Estimate a few hundred bytes per score event (obviously, size depends on the number of parameters) and avoid using all of your computer's memory.
The `sf-granulate` function is used for granular synthesis. It takes a sound file named `filename` as the source for grains. Grains are extracted from the sound file by stepping through the file in equal increments. Each grain duration is the sum of `grain-dur` and a random number from 0 to `grain-dev`. Grains are then multiplied by a raised cosine smoothing window and resampled at a ratio between 1.0 and `pitch-dev`. If `pitch-dev` is greater than one, grains are stretched and the pitch (if any) goes down. If `pitch-dev` is less than one, grains are shortened and the pitch goes up. Grains are then output with an inter-onset interval between successive grains (which may overlap) determined by the sum of `ioi` and a random number from 0 to `ioi-dev`. The duration of the resulting sound is determined by the stretch factor (not by the sound file). The number of grains is the total sound duration (determined by the stretch factor) divided by the mean inter-onset interval, which is \( \text{ioi} + \text{ioi-dev} \times 0.5 \). The grains are taken from equally-spaced starting points in `filename`, and depending on grain size and number, the grains may or may not overlap. The output duration will simply be the sum of the inter-onset intervals and the duration of the last grain. If `ioi-dev` is non-zero, the output duration will vary, but the expected value of the duration is the stretch factor. To achieve a rich granular synthesis effect, it is often a good idea to sum four or more copies of `sf-granulate` together. (See the `gran-test` function in `gran.lsp`.)

### 15.14 Chowning FM Voices

John Chowning developed voice synthesis methods using FM to simulate resonances for his 1981 composition "Phone." He later recreated the synthesis algorithms in Max, and Jorge Sastre ported these to SAL. See `demos/FM-voices-Chowning.sal` for more details.

### 15.15 Atonal Melody Composition

Jorge Sastre contributed `demos/atonal-melodies.sal`, code that generates atonal melodies. You can find links to an example score and audio file in the code and also at [http://algocompbook.com/examples.html](http://algocompbook.com/examples.html).

### 15.16 MIDI Utilities

The `midishow.lsp` library has functions that can print the contents fo MIDI files. This intended as a debugging aid.

```lisp
(midi-show-file file-name) [SAL]
(midi-show-file file-name) [LISP]
  Print the contents of a MIDI file to the console.

(midi-show the-seq [out-file]) [SAL]
(midi-show the-seq [out-file]) [LISP]
  Print the contents of the sequence `the-seq` to the file `out-file` (whose default value is the console.)
```
15.17 Reverberation

The reverb.lsp library implements artificial reverberation.

\[\text{reverb}(\text{snd}, \text{time}) \text{ [SAL]}\]
\[(\text{reverb} \text{snd time}) \text{ [LISP]}\]
Artificial reverberation applied to \text{snd} with a decay time of \text{time}.

15.18 DTMF Encoding

The dtmf.lsp library implements DTMF encoding. DTMF is the "touch tone" code used by telephones.

\[\text{dtmf-tone}(\text{key}, \text{len}, \text{space}) \text{ [SAL]}\]
\[(\text{dtmf-tone} \text{key len space}) \text{ [LISP]}\]
Generate a single DTMF tone. The \text{key} parameter is either a digit (a FIXNUM from 0 through 9) or the atom STAR or POUND. The duration of the done is given by \text{len} (a FLONUM) and the tone is followed by silence of duration \text{space} (a FLONUM).

\[\text{speed-dial}(\text{thelist}) \text{ [SAL]}\]
\[(\text{speed-dial thelist}) \text{ [LISP]}\]
Generates a sequence of DTMF tones using the keys in \text{thelist} (a LIST of keys as described above under \text{dtmf-tone}). The duration of each tone is 0.2 seconds, and the space between tones is 0.1 second. Use \text{stretch} to change the "dialing" speed.

15.19 Dolby Surround(R), Stereo and Spatialization Effects

The spatial.lsp library implements various functions for stereo manipulation and spatialization. It also includes some functions for Dolby Pro-Logic panning, which encodes left, right, center, and surround channels into stereo. The stereo signal can then be played through a Dolby decoder to drive a surround speaker array. This library has a somewhat simplified encoder, so you should certainly test the output. Consider using a high-end encoder for critical work. There are a number of functions in spatial.lsp for testing. See the source code for comments about these.

\[\text{stereoize}(\text{snd}) \text{ [SAL]}\]
\[(\text{stereoize snd}) \text{ [LISP]}\]
Convert a mono sound, \text{snd}, to stereo. Four bands of equalization and some delay are used to create a stereo effect.

\[\text{widen}(\text{snd}, \text{amt}) \text{ [SAL]}\]
\[(\text{widen snd amt}) \text{ [LISP]}\]
Artificially widen the stereo field in \text{snd}, a two-channel sound. The amount of widening is \text{amt}, which varies from 0 (\text{snd} is unchanged) to 1 (maximum widening). The \text{amt} can be a SOUND or a number.
span(snd, amt) [SAL]  
(span snd amt) [LISP]
    Pan the virtual center channel of a stereo sound, snd, by amt, where 0 pans all the way to the left, while 1 pans all the way to the right. The amt can be a SOUND or a number.

swapchannels(snd) [SAL]  
(swapchannels snd) [LISP]
    Swap left and right channels in snd, a stereo sound.

prologic(l, c, r, s) [SAL]  
(prologic l c r s) [LISP]
    Encode four monaural SOUNDs representing the front-left, front-center, front-right, and rear channels, respectively. The return value is a stereo sound, which is a Dolby-encoded mix of the four input sounds.

pl-left(snd) [SAL]  
(pl-left snd) [LISP]
    Produce a Dolby-encoded (stereo) signal with snd, a SOUND, encoded as the front left channel.

pl-center(snd) [SAL]  
(pl-center snd) [LISP]
    Produce a Dolby-encoded (stereo) signal with snd, a SOUND, encoded as the front center channel.

pl-right(snd) [SAL]  
(pl-right snd) [LISP]
    Produce a Dolby-encoded (stereo) signal with snd, a SOUND, encoded as the front right channel.

pl-rear(snd) [SAL]  
(pl-rear snd) [LISP]
    Produce a Dolby-encoded (stereo) signal with snd, a SOUND, encoded as the rear, or surround, channel.

pl-pan2d(snd, x, y) [SAL]  
(pl-pan2d snd x y) [LISP]
    Comparable to Nyquist’s existing pan function, pl-pan2d provides not only left-to-right panning, but front-to-back panning as well. The function accepts three parameters: snd is the (monophonic) input SOUND, x is a left-to-right position, and y is a front-to-back position. Both position parameters may be numbers or SOUNDs. An x value of 0 means left, and 1 means right. Intermediate values map linearly between these extremes. Similarly, a y value of 0 causes the sound to play entirely through the front speakers(s), while 1 causes it to play entirely through the rear. Intermediate values map linearly. Note that, although there are usually two rear speakers in Pro-Logic systems, they are both driven by the same signal. Therefore any sound that is panned totally to the rear will be played over both rear speakers. For example, it is not possible to play a sound exclusively through the rear left speaker.

pl-position(snd, x, y, config) [SAL]  
(pl-position snd x y config) [LISP]
    The position function builds upon speaker panning to allow more abstract placement of sounds. Like pl-pan2d, it accepts a (monaural) input sound as well as left-to-right (x) and front-to-back (y) coordinates, which may be FLONUMs or SOUNDs. A fourth parameter config specifies the distance from listeners to the speakers (in meters). Current settings assume this to be constant for all speakers, but this assumption can be changed easily (see comments in the code for more detail). There are several important differences between pl-position and pl-pan2d. First, pl-position uses a Cartesian coordinate system that allows x and y coordinates outside of the range (0, 1). This model assumes a listener position of (0,0). Each speaker has a predefined position as well. The input sound’s position, relative to the listener, is given by the vector (x,y).

pl-doppler(snd, r) [SAL]
(pl-doppler snd r) [LISP]
Pitch-shift moving sounds according to the equation: \( fr = f0((c+vr)/c) \), where \( fr \) is the output frequency, \( f0 \) is the emitted (source) frequency, \( c \) is the speed of sound (assumed to be 344.31 m/s), and \( vr \) is the speed at which the emitter approaches the receiver. \( vr \) is the first derivative of parameter \( r \), the distance from the listener in meters.

15.20  Drum Machine

The drum machine software in demos/plight deserves further explanation. to use the software, load the code by evaluating:

```
load "../demos/plight/drum.lsp"
exec load-props-file(strcat(*plight-drum-path*, "beats.props"))
exec create-drum-patches()
exec create-patterns()
```

Drum sounds and patterns are specified in the beats.props file (or whatever name you give to load-props-file). This file contains two types of specifications. First, there are sound file specifications. Sound files are located by a line of the form:

```
set sound-directory = "kit/
```

This gives the name of the sound file directory, relative to the beats.props file. Then, for each sound file, there should be a line of the form:

```
track.2.5 = big-tom-5.wav
```

This says that on track 2, a velocity value of 5 means to play the sound file big-tom-5.wav. (Tracks and velocity values are described below.) The beats.props file contains specifications for all the sound files in demos/plight/kit using 8 tracks. If you make your own specifications file, tracks should be numbered consecutively from 1, and velocities should be in the range of 1 to 9.

The second set of specifications is of beat patterns. A beat pattern is given by a line in the following form:

```
beats.5 = 2--32--43-4-5---
```

The number after beats is just a pattern number. Each pattern is given a unique number. After the equal sign, the digits and dashes are velocity values where a dash means “no sound.” Beat patterns should be numbered consecutively from 1.

Once data is loaded, there are several functions to access drum patterns and create drum sounds (described below). The demos/plight/drums.lsp file contains an example function plight-drum-example to play some drums. There is also the file demos/plight/beats.props to serve as an example of how to specify sound files and beat patterns.

```
drum(tracknum, patternnum, bpm) [SAL]
(drum tracknum patternnum bpm) [LISP]
Create a sound by playing drums sounds associated with track tracknum (a FIXNUM) using pattern patternnum. The tempo is given by bpm in beats per minute. Normally patterns are a sequence of sixteenth notes, so the tempo is in sixteenth notes per minute. For example, if patternnum is 10, then use the pattern specified for beats.10. If the third character of this pattern is 3 and tracknum is 5, then on the third beat, play the soundfile assigned to track.5.3. This function returns a SOUND.
```
drum-loop(snd, duration, numtimes) [SAL]
(drum-loop snd duration numtimes) [LISP]
  Repeat the sound given by snd numtimes times. The repetitions occur at a time offset of duration, regardless of
  the actual duration of snd. A SOUND is returned.

length-of-beat(bpm) [SAL]
(length-of-beat bpm) [LISP]
  Given a tempo of bpm, return the duration of the beat in seconds. Note that this software has no real notion of
  beat. A “beat” is just the duration of each character in the beat pattern strings. This function returns a FLONUM.

# 15.21 Minimoog-inspired Synthesis

The moog.lsp library gives the Nyquist user easy access to "classic" synthesizer sounds through an emulation of
the Minimoog Synthesizer. Unlike modular Moogs that were very large, the Minimoog was the first successful and
commonly used portable synthesizer. The trademark filter attack was unique and easily recognizable. The goal of this
Nyquist instrument is not only to provide the user with default sounds, but also to give control over many of the "knobs"
found on the Minimoog. In this implementation, these parameters are controlled using keywords. The input to the moog
instrument is a user-defined sequence of notes, durations, and articulations that simulate notes played on a keyboard.
These are translated into control voltages that drive multiple oscillators, similar to the Voltage Controlled Oscillator or
VCO found in the original analog Moog.

The most recognizable feature of the Minimoog is its resonant filter, a Four-Pole Ladder Filter invented by Robert
Moog. It is simply implemented in a circuit with four transistors and provides an outstanding 24 dB/octave rolloff. It
is modeled here using the built-in Nyquist resonant filter. One of the Moog filter features is a constant Q, or center
frequency to bandwidth ratio. This is implemented and the user can control the Q.

The user can control many parameters using keywords. Their default values, acceptable ranges, and descriptions are
shown below. The defaults were obtained by experimenting with the official Minimoog software synthesizer by Arturia.

## 15.21.1 Oscillator Parameters

range-osc1(2)
range-osc2(1)
range-osc3(3)

These parameters control the octave of each oscillator. A value of 1 corresponds to the octave indicated by the input
note. A value of 3 is two octaves above the fundamental. The allowable range is 1 to 7.

detun2(-.035861)
detun3(.0768)

Detuning of two oscillators adds depth to the sound. A value of 1 corresponds to an increase of a single semitone and a
-1 corresponds to a decrease in a semitone. The range is -1 to 1.
Figure 15.1: System diagram for Minimoog emulator.

\begin{verbatim}
shape-osc1(*saw-table*)
shape-osc2(*saw-table*)
shape-osc3(*saw-table*)
\end{verbatim}

Oscillators can use any wave shape. The default sawtooth waveform is a built-in Nyquist variable. Other waveforms can be defined by the user.

\begin{verbatim}
volume-osc1(1)
volume-osc2(1)
volume-osc3(1)
\end{verbatim}

These parameters control the relative volume of each oscillator. The range is any FLONUM greater than or equal to zero.

### 15.21.2 Noise Parameters

\begin{verbatim}
noiselevel(.05)
\end{verbatim}

This parameter controls the relative volume of the noise source. The range is any FLONUM greater than or equal to zero.

### 15.21.3 Filter Parameters

\begin{verbatim}
filter-cutoff(768)
\end{verbatim}
The cutoff frequency of the filter is given in Hz. The range is zero to 20,000 Hz.

\[ Q(2) \]

\( Q \) is the ratio of center frequency to bandwidth. It is held constant by making the bandwidth a function of frequency. The range is any \( \text{FLONUM} \) greater than zero.

\[ \text{contour(.65)} \]

Contour controls the range of the transient frequency sweep from a high to low cutoff frequency when a note is played. The high frequency is proportional to contour. A contour of 0 removes this sweep. The range is 0 to 1.

\[ \text{filter-attack(.0001)} \]

Filter attack controls the attack time of the filter, i.e. the time to reach the high cutoff frequency. The range is any \( \text{FLONUM} \) greater than zero (seconds).

\[ \text{filter-decay(.5)} \]

Filter decay controls the decay time of the filter, i.e. the time of the sweep from the high to low cutoff frequency. The range is any \( \text{FLONUM} \) greater than zero (seconds).

\[ \text{filter-sustain(.8)} \]

Filter sustain controls the percentage of the filter cutoff frequency that the filter settles on following the sweep. The range is 0 to 1.

### 15.21.4 Amplitude Parameters

\[ \text{amp-attack(.01)} \]

This parameter controls the amplitude envelope attack time, i.e. the time to reach maximum amplitude. The range is any \( \text{FLONUM} \) greater than zero (seconds).

\[ \text{amp-decay(1)} \]

This parameter controls the amplitude envelope decay time, i.e. the time between the maximum and sustain volumes. The range is any \( \text{FLONUM} \) greater than zero (seconds).

\[ \text{amp-sustain(1)} \]

This parameter controls the amplitude envelope sustain volume, a fraction of the maximum. The range is 0 to 1.

\[ \text{amp-release(0)} \]

This parameter controls the amplitude envelope release time, i.e. the time it takes between the sustain volume and 0 once the note ends. The duration controls the overall length of the sound. The range of \( \text{amp-release} \) is any \( \text{FLONUM} \) greater than zero (seconds).
15.21.5 Other Parameters

`glide(0)`

Glide controls the low-pass filter on the control voltages. This models the glide knob on a Minimoog. A higher value corresponds to a lower cutoff frequency and hence a longer "glide" between notes. A value of 0 corresponds to no glide. The range is zero to 10.

15.21.6 Input Format

A single note or a series of notes can be input to the Moog instrument by defining a list with the following format:

```
list(list(frequency, duration, articulation), ...
```

where `frequency` is a `FLONUM` in steps, `duration` is the duration of each note in seconds (regardless of the release time of the amplifier), and `articulation` is a percentage of the duration that a sound will be played, representing the amount of time that a key is pressed. The filter and amplitude envelopes are only triggered if a note is played when the articulation of the previous note is less than 1, or a key is not down at the same time. This Moog instrument is a monophonic instrument, so only one note can sound at a time. The release section of the amplifier is triggered when the articulation is less than 1 at the time (`duration * articulation`).

15.21.7 Sample Code/Sounds

**Sound 1 (default parameters):**

```scheme
set s = {{24 .5 .99} {26 .5 .99} {28 .5 .99}
         {29 .5 .99} {31 2 1}}
play moog(s)
```

**Sound 2 (articulation, with amplitude release):**

```scheme
set s = {{24 .5 .5} {26 .5 1} {28 .5 .25} {29 .5 1} {31 1 .8}}
play moog(s, amp-release: .2)
```

**Sound 3 (glide):**

```scheme
set s = {{24 .5 .5} {38 .5 1} {40 .5 .25}
         {53 .5 1} {55 2 1} {31 2 .8} {36 2 .8}}
play moog(s, amp-release: .2, glide: .5)
```

**Sound 4 (keyword parameters):** Filter attack and decay are purposely longer than notes being played with articulation equal to 1.

```scheme
set s = {{20 .5 1} {27 .5 1} {26 .5 1} {21 .5 1}
         {20 .5 1} {27 .5 1} {26 .5 1} {21 .5 1}}
play moog(s, shape-osc1: *tri-table*, shape-osc2: *tri-table*,
         filter-attack: 2, filter-decay: 2,
         filter-cutoff: 300, contour: .8, glide: .2, Q: 8)
```

**Sound 5:** This example illustrates the ability to completely define a new synthesizer with different parameters creating a drastically different sound. Sine waves are used for wavetables. There is a high value for glide.
define function my-moog(freq)
    return moog(freq,
        range-osc1: 3, range-osc2: 2, range-osc3: 4,
        detun2: -.043155, detun3: .015016,
        noiselevel: 0,
        filter-cutoff: 400, Q: .1, contour: .0000001,
        filter-attack: 0, filter-decay: .01, filter-sustain: 1,
        shape-osc1: *sine-table*, shape-osc2: *sine-table*,
        shape-osc3: *sine-table*, volume-osc1: 1, volume-osc2: 1,
        volume-osc3: 1, amp-attack: .1, amp-decay: 0,
        amp-sustain: 1, amp-release: .3, glide: 2)

set s = {{80 .4 .75} {28 .2 1} {70 .5 1} {38 1 .5}}
play my-moog(s)

Sound 6: This example has another variation on the default parameters.

set s = {{24 .5 .99} {26 .5 .99} {28 .5 .99}
    {29 .5 .99} {31 2 1}}
play moog(s, shape-osc1: *tri-table*, shape-osc2: *tri-table*,
    filter-attack: .5, contour: .5)
Bibliography


Appendix A  Extending Nyquist

**WARNING:** Nyquist sound functions look almost like a human wrote them; they even have a fair number of comments for human readers. Don’t be fooled: virtually all Nyquist functions are written by a special translator. If you try to write a new function by hand, you will probably not succeed, and even if you do, you will waste a great deal of time. (End of Warning.)

### A.1 Translating Descriptions to C Code

The translator code used to extend Nyquist resides in the `trnsrc` directory. This directory also contains a special `init.lsp`, so if you start XLisp or Nyquist in this directory, it will automatically read `init.lsp`, which in turn will load the translator code (which resides in several files).

Also in the `trnsrc` directory are a number of `.alg` files, which contain the source code for the translator (more on these will follow), and a number of corresponding `.h` and `.c` files.

To translate a `.alg` file to `.c` and `.h` files, you start XLisp or Nyquist in the `trnsrc` directory and type

```
(translate "prod")
```

where "prod" should really be replaced by the filename (without a suffix) you want to translate. Be sure you have a saved, working copy of Nyquist or XLisp before you recompile!

**Note:** On the Macintosh, just run Nyquist out of the `runtime` directory and then use the Load menu command to load `init.lsp` from the `trnsrc` directory. This will load the translation code and change Nyquist’s current directory to `trnsrc` so that commands like `(translate "prod")` will work.

### A.2 Rebuilding Nyquist

After generating `prod.c` and `prod.h`, you need to recompile Nyquist. For Unix systems, you will want to generate a new Makefile. Modify `transfiles.lsp` in your main Nyquist directory, run XLisp or Nyquist and load `makefile.lsp`. Follow the instructions to set your machine type, etc., and execute `(makesrc)` and `(makefile)`. 

210
A.3 Accessing the New Function

The new Lisp function will generally be named with a snd- prefix, e.g. snd-prod. You can test this by running Nyquist. Debugging is usually a combination of calling the code from within the interpreter, reading the generated code when things go wrong, and using a C debugger to step through the inner loop of the generated code. An approach I like is to set the default sample rate to 10 hertz. Then, a one-second sound has only 10 samples, which are easy to print and study on a text console.

For some functions, you must write some Lisp code to impose ordinary Nyquist behaviors such as stretching and time shifting. A good approach is to find some structurally similar functions and see how they are implemented. Most of the Lisp code for Nyquist is in nyquist.lsp.

Finally, do not forget to write up some documentation. Also, contributions are welcome. Send your .alg file, documentation, Lisp support functions for nyquist.lsp, and examples or test programs to rbd@cs.cmu.edu. I will either put them in the next release or make them available at a public ftp site.

A.4 Why Translation?

Many of the Nyquist signal processing operations are similar in form, but they differ in details. This code is complicated by many factors: Nyquist uses lazy evaluation, so the operator must check to see that input samples are available before trying to access them. Nyquist signals can have different sample rates, different block sizes, different block boundaries, and different start times, all of which must be taken into account. The number of software tests is enormous. (This may sound like a lot of overhead, but the overhead is amortized over many iterations of the inner loop. Of course setting up the inner loop to run efficiently is one more programming task.)

The main idea behind the translation is that all of the checks and setup code are similar and relatively easy to generate automatically. Programmers often use macros for this sort of task, but the C macro processor is too limited for the complex translation required here. To tell the translator how to generate code, you write .alg files, which provide many details about the operation in a declarative style. For example, the code generator can make some optimizations if you declare that two input signals are commutative (they can be exchanged with one another). The main part of the .alg file is the inner loop which is the heart of the signal processing code.

A.5 Writing a .alg File

WARNING: Translation relies heavily on string substitution, which is fragile. In particular, variables with names that are substrings of other variables will cause problems. For example if you declare STATE variables "phase" and "iphase", then the translator will globally substitute "phase_reg" for "phase", converting "phase" to "phase_reg" and iphase" to "iphase_reg". Then it will substitute "iphase_reg" for iphase" which will convert the existing "iphase_reg" to "iphase_reg_reg". This will be confusing and will not compile. (End of WARNING)

To give you some idea how functions are specified, here is the specification for snd-prod, which generates over 250 lines of C code:
(PROD-ALG
  (NAME "prod")
  (ARGUMENTS ("sound_type" "s1") ("sound_type" "s2"))
  (START (MAX s1 s2))
  (COMMUTATIVE (s1 s2))
  (INNER-LOOP "output = s1 * s2")
  (LINEAR s1 s2)
  (TERMINATE (MIN s1 s2))
  (LOGICAL-STOP (MIN s1 s2))
)

A .alg file is always of the form:

(name
  (attribute value)
  (attribute value)
  ...
)

There should be just one of these algorithms descriptions per file. The name field is arbitrary: it is a Lisp symbol whose property list is used to save the following attribute/value pairs. There are many attributes described below. For more examples, see the .alg files in the trnsrc directory.

Understanding what the attributes do is not easy, so here are three recommendations for implementors. First, if there is an existing Nyquist operator that is structurally similar to something you want to implement, make a copy of the corresponding .alg file and work from there. In some cases, you can merely rename the parameters and substitute a new inner loop. Second, read the generated code, especially the generated inner loop. It may not all make sense, but sometimes you can spot obvious errors and work your way back to the error in the .alg file. Third, if you know where something bad is generated, see if you can find where the code is generated. (The code generator files are listed in init.lsp.) This code is poorly written and poorly documented, but in some cases it is fairly straightforward to determine what attribute in the .alg file is responsible for the erroneous output.

### A.6 Attributes

Here are the attributes used for code generation. Attributes and values may be specified in any order.

(NAME "string") specifies a base name for many identifiers. In particular, the generated filenames will be string.c and string.h, and the XLisp function generated will be snd-string.

(ARGUMENTS arglist) describes the arguments to be passed from XLisp. Arglist has the form: (type1 name1) (type2 name2) ..., where type and name are strings in double quotes, e.g. ("sound_type" "s") specifies a SOUND parameter named s. Note that arglist is not surrounded by parentheses. As seen in this example, the type names and parameter names are C identifiers. Since the parameters are passed in from XLisp, they must be chosen from a restricted set. Valid type names are: "sound_type", "rate_type", "double", "long", "string", and "LVAL".
(STATE statelist)
  describes additional state (variables) needed to perform the computation. A statelist is similar to an arglist (see ARGUMENTS above), and has the form: (type1 name1 init1 [TEMP]) (type2 name2 init2 [TEMP])
  ... The types and names are as in arglist, and the "inits" are double-quoted initial values. Initial values may be any C expression. State is initialized in the order implied by statelist when the operation is first called from XLisp. If TEMP is omitted the state is preserved in a structure until the sound computation completes. Otherwise, the state variable only exists at state initialization time.

(INNER-LOOP innerloop-code)
  describes the inner loop, written as C code. The innerloop-code is in double quotes, and may extend over multiple lines. To make generated code extra-beautiful, prefix each line of innerloop-code with 12 spaces. Temporary variables should not be declared at the beginning of innerloop-code. Use the INNER-LOOP-LOCALS attribute instead. Within innerloop-code, each ARGUMENT of type sound_type must be referenced exactly one time. If you need to use a signal value twice, assign it once to a temporary and use the temporary twice. The inner loop must also assign one time to the pseudo-variable output. The model here is that the name of a sound argument denotes the value of the corresponding signal at the current output sample time. The inner loop code will be called once for each output sample. In practice, the code generator will substitute some expression for each signal name. For example, prod.alg specifies

  (INNER-LOOP "output = s1 * s2")

(s1 and s2 are ARGUMENTS.) This expands to the following inner loop in prod.c:

  *out_ptr_reg++ = *s1_ptr_reg++ * *s2_ptr_reg++;

In cases where arguments have different sample rates, sample interpolation is in-lined, and the expressions can get very complex. The translator is currently very simple-minded about substituting access code in the place of parameter names, and this is a frequent source of bugs. Simple string substitution is performed, so you must not use a parameter or state name that is a substring of another. For example, if two sound parameters were named s and s2, the translator might substitute for "s" in two places rather than one. If this problem occurs, you will almost certainly get a C compiler syntax error. The fix is to use "more unique" parameter and state variable names.

(INNER-LOOP-LOCALS "innerloop-code")
  The innerloop-code contains C declarations of local variables set and referenced in the inner loop.

(SAMPLE-RATE "expr")
  specifies the output sample rate; expr can be any C expression, including a parameter from the ARGUMENTS list. You can also write (SAMPLE-RATE (MAX name1 name2 ...)) where names are unquoted names of arguments.

(SUPPORT-HEADER "c-code")
  specifies arbitrary C code to be inserted in the generated .h file. The code typically contains auxiliary function declarations and definitions of constants.

(SUPPORT-FUNCTIONS "c-code")
  specifies arbitrary C code to be inserted in the generated .c file. The code typically contains auxiliary functions and definitions of constants.
(FINALIZATION "c-code")
specifies code to execute when the sound has been fully computed and the state variables are about to be
decallocated. This is the place to deallocate buffer memory, etc.

(CONSTANT "name1" "name2" ...)  
specifies state variables that do not change value in the inner loop. The values of state variables are loaded into
registers before entering the inner loop so that access will be fast within the loop. On exiting the inner loop, the
final register values are preserved in a "suspension" structure. If state values do not change in the inner loop, this
CONSTANT declaration can eliminate the overhead of storing these registers.

(START spec)
specifies when the output sound should start (a sound is zero and no processing is done before the start time).
The spec can take several forms: (MIN name1 name2 ...) means the start time is the minimum of the start
times of input signals name1, name2, .... Note that these names are not quoted.

(TERMINATE spec)
specifies when the output sound terminates (a sound is zero after this termination time and no more samples
are computed). The spec can take several forms: (MIN name1 name2 ...) means the terminate time is the
minimum of the terminate times of input arguments name1, name2, .... Note that these names are not quoted.
To terminate at the time of a single argument s1, specify (MIN s1). To terminate after a specific duration, use
(AFTER "c-expr"), where c-expr is a C variable or expression. To terminate at a particular time, use (AT
"c-expr"). spec may also be COMPUTED, which means to use the maximum sample rate of any input signal.

(LOGICAL-STOP spec)
specifies the logical stop time of the output sound. This spec is just like the one for TERMINATE. If no
LOGICAL-STOP attribute is present, the logical stop will coincide with the terminate time.

(ALWAYS-SCALE name1 name2 ...)  
says that the named sound arguments (not in quotes) should always be multiplied by a scale factor. This is a
space-time tradeoff. When Nyquist sounds are scaled, the scale factor is merely stored in a structure. It is the
responsibility of the user of the samples to actually scale them (unless the scale factor is exactly 1.0). The default
is to generate code with and without scaling and to select the appropriate code at run time. If there are N signal
inputs, this will generate 2^N versions of the code. To avoid this code explosion, use the ALWAYS-SCALE attribute.

(INLINE-INTERPOLATION flag)
controls when sample rate interpolation should be performed in-line in the inner loop. There are two forms
of sample rate interpolation. One is intended for use when the rate change is large and many points will be
interpolated. This form uses a divide instruction and some setup at the low sample rate, but the inner loop
overhead is just an add. The other form, intended for less drastic sample rate changes, performs interpolation
with 2 multiplies and several adds per sample at the high sample rate. If inline interpolation is enabled, Nyquist
generates various inner loops and selects the appropriate one at run-time. (This can cause a combinatorial
explosion if there are multiple sound arguments.) If inline interpolation is not enabled, much less code is
generated and interpolation is performed as necessary by instantiating a separate signal processing operation. The
value of flag is YES to generate inline interpolation, NO to disable inline interpolation, and NIL to take the default
set by the global variable *INLINE-INTERPOLATION*. The default is also taken if no INLINE-INTERPOLATION
attribute is specified.

(STEP-FUNCTION name1 name2 ...)
Normally all argument signals are linearly interpolated to the output sample rate. The linear interpolation can be
turned off with this attribute. This is used, for example, in Nyquist variable filters so that filter coefficients are
computed at low sample rates. In fact, this attribute was added for the special case of filters.
(DEPENDS spec1 spec2 ...)
Specifies dependencies. This attribute was also introduced to handle the case of filter coefficients (but may have other applications.) Use it when a state variable is a function of a potentially low-sample-rate input where the input is in the STEP-FUNCTION list. Consider a filter coefficient that depends upon an input signal such as bandwidth. In this case, you want to compute the filter coefficient only when the input signal changes rather than every output sample, since output may occur at a much higher sample rate. A spec is of the form

\[
(\text{"name" } \text{"arg" } \text{"expr" [TEMP "type" ]})
\]

which is interpreted as follows: name depends upon arg; when arg changes, recompute expr and assign it to name. The name must be declared as a STATE variable unless TEMP is present, in which case name is not preserved and is used only to compute other state. Variables are updated in the order of the DEPENDS list.

(FORCE-INTO-REGISTER name1 name2 ...)
causes name1, name2, ... to be loaded into registers before entering the inner loop. If the inner loop references a state variable or argument, this happens automatically. Use this attribute only if references are "hidden" in a #define'd macro or referenced in a DEPENDS specification.

(NOT-REGISTER name1 name2 ...)
specifies state and arguments that should not be loaded into registers before entering an inner loop. This is sometimes an optimization for infrequently accessed state.

(NOT-IN-INNER-LOOP "name1" "name2" ...)
says that certain arguments are not used in the inner loop. Nyquist assumes all arguments are used in the inner loop, so specify them here if not. For example, tables are passed into functions as sounds, but these sounds are not read sample-by-sample in the inner loop, so they should be listed here.

(MAINTAIN ("name1" "expr1") ("name2" "expr2") ...)
Sometimes the IBM XLC compiler generates better loop code if a variable referenced in the loop is not referenced outside of the loop after the loop exit. Technically, optimization is better when variables are dead upon loop exit. Sometimes, there is an efficient way to compute the final value of a state variable without actually referencing it, in which case the variable and the computation method are given as a pair in the MAINTAIN attribute. This suppresses a store of the value of the named variable, making it a dead variable. Where the store would have been, the expression is computed and assigned to the named variable. See partial.alg for an example. This optimization is never necessary and is only for fine-tuning.

(LINEAR name1 name2 ...)
specifies that named arguments (without quotes) are linear with respect to the output. What this really means is that it is numerically OK to eliminate a scale factor from the named argument and store it in the output sound descriptor, avoiding a potential multiply in this inner loop. For example, both arguments to snd-prod (signal multiplication) are "linear." The inner loop has a single multiplication operator to multiply samples vs. a potential 3 multiplies if each sample were also scaled. To handle scale factors on the input signals, the scale factors are automatically multiplied and the product becomes the scale factor of the resulting output. (This effectively "passes the buck" to some other, or perhaps more than one, signal processing function, which is not always optimal. On the other hand, it works great if you multiply a number of scaled signals together: all the scale factors are ultimately handled with a single multiply.)

(INTERNAL-SCALING name1 name2 ...)
indicates that scaling is handled in code that is hidden from the code generator for name1, name2, ..., which are sound arguments. Although it is the responsibility of the reader of samples to apply any given scale factor, sometimes scaling can be had for free. For example, the snd-recip operation computes the reciprocal of the input samples by preparing a division. The simple approach would be to specify an inner loop of output = 1.0/a1, where a1 is the input. With scaling, this would generate an inner loop something like this:
*output++ = 1.0 / (s1_scale_factor * *s1++);  

but a much better approach would be the following:

*output++ = my_scale_factor / *s1++

where my_scale_factor is initialized to 1.0 / s1->scale. Working backward from the desired inner loop to the .alg inner loop specification, a first attempt might be to specify:

(INNER-LOOP "output = my_scale_factor / s1")

but this will generate the following:

*output++=my_scale_factor/(s1_scale_factor * *s1++);

Since the code generator does not know that scaling is handled elsewhere, the scaling is done twice! The solution is to put s1 in the INTERNAL-SCALING list, which essentially means "I've already incorporated scaling into the algorithm, so suppress the multiplication by a scale factor."

(COMMUTATIVE (name1 name2 ...))

specifies that the results will not be affected by interchanging any of the listed arguments. When arguments are commutative, Nyquist rearranges them at run-time into decreasing order of sample rates. If interpolation is in-line, this can dramatically reduce the amount of code generated to handle all the different cases. The prime example is prod.alg.

(TYPE-CHECK "code")

specifies checking code to be inserted after argument type checking at initialization time. See downproto.alg for an example where a check is made to guarantee that the output sample rate is not greater than the input sample rate. Otherwise an error is raised.

A.7 Generated Names

The resulting .c file defines a number of procedures. The procedures that do actual sample computation are named something like name_interp-spec_FETCH, where name is the NAME attribute from the .alg file, and interp-spec is an interpolation specification composed of a string of the following letters: n, s, i, and r. One letter corresponds to each sound argument, indicating no interpolation (r), scaling only (s), ordinary linear interpolation with scaling (i), and ramp (incremental) interpolation with scaling (r). The code generator determines all the combinations of n, s, i, and r that are necessary and generates a separate fetch function for each.

Another function is name_toss_fetch, which is called when sounds are not time-aligned and some initial samples must be discarded from one or more inputs.

The function that creates a sound is snd_make_name. This is where state allocation and initialization takes place. The proper fetch function is selected based on the sample rates and scale factors of the sound arguments, and a sound_type is returned.

Since Nyquist is a functional language, sound operations are not normally allowed to modify their arguments through side effects, but even reading samples from a sound_type causes side effects. To hide these from the Nyquist programmer, sound_type arguments are first copied (this only copies a small structure. The samples themselves are on a shared list). The function snd_name performs the necessary copies and calls snd_make_name. It is the snd_name function that is called by XLisp. The XLisp name for the function is SND-NAME. Notice that the underscore in C is
converted to a dash in XLisp. Also, XLisp converts identifiers to upper case when they are read, so normally, you would type `snd-name` to call the function.

### A.8 Scalar Arguments

If you want the option of passing either a number (scalar) or a signal as one of the arguments, you have two choices, neither of which is automated. Choice 1 is to coerce the constant into a signal from within XLisp. The naming convention would be to `DEFUN` a new function named `NAME` or `S-NAME` for ordinary use. The `NAME` function tests the arguments using XLisp functions such as `TYPE-OF`, `NUMBERP`, and `SOUNDP`. Any number is converted to a `SOUND`, e.g. using `CONST`. Then `SND-NAME` is called with all sound arguments. The disadvantage of this scheme is that scalars are expanded into a sample stream, which is slower than having a special inner loop where the scalar is simply kept in a register, avoiding loads, stores, and addressing overhead.

Choice 2 is to generate a different sound operator for each case. The naming convention here is to append a string of c’s and v’s, indicating constant (scalar) or variable (signal) inputs. For example, the `reson` operator comes in four variations: `reson`, `resoncv`, `resonvc`, and `resonvv`. The `resonvc` version implements a resonating filter with a variable center frequency (a sound type) and a constant bandwidth (a `FLONUM`). The `RESON` function in Nyquist is an ordinary Lisp function that checks types and calls one of `SND-RESON`, `SND-RESONCV`, `SND-RESONVC`, or `SND-RESONVV`.

Since each of these `SND-` functions performs further selection of implementation based on sample rates and the need for scaling, there are 25 different functions for computing `RESON`! So far, however, Nyquist is smaller than Common Lisp and it’s about half the size of Microsoft Word. Hopefully, exponential growth in memory density will outpace linear (as a function of programming effort) growth of Nyquist.
Appendix B   Intgen

This documentation describes Intgen, a program for generating XLISP to C interfaces. Intgen works by scanning .h files with special comments in them. Intgen builds stubs that implement XLISP SUBR’s. When the SUBR is called, arguments are type-checked and passed to the C routine declared in the .h file. Results are converted into the appropriate XLISP type and returned to the calling XLISP function. Intgen lets you add C functions into the XLISP environment with very little effort.

B.1 Overview

The interface generator will take as command-line input:

- the name of the .c file to generate (do not include the .c extension; e.g. write xlexten, not xlexten.c);
- a list of .h files.

Alternatively, the command line may specify a command file from which to read file names. The command file name should be preceded by "@", for example:

```bash
intgen @sndfns.cl
```

reads sndfns.cl to get the command-line input. Only one level of indirection is allowed.

The output is:

- a single .c file with one SUBR defined for each designated routine in a .h file.
- a .h file that declares each new C routine. E.g. if the .c file is named xlexten.c, this file will be named xlextendedefs.h;
- a .h file that extends the SUBR table used by Xlisp. E.g. if the .c file is named xlexten.c, then this file is named xlextenptrs.h;
- a .lsp file with lisp initialization expressions copied from the .h files. This file is only generated if at least one initialization expression is encountered.

For example, the command line

```bash
intgen seint "setypes.h access.h"
```
generates the file seint.c, using declarations in setypes.h and access.h. Normally, the .h files are included by the generated file using #include commands. A ~ before a file means do not include the .h file. (This may be useful if you extend xlisp.h, which will be included anyway). Also generated will be setintdefs.h and seintptrs.h.

### B.1.1 Extending Xlisp

Any number of .h files may be named on the command line to Intgen, and Intgen will make a single .c file with interface routines for all of the .h files. On the other hand, it is not necessary to put all of the extensions to Xlisp into a single interface file. For example, you can run Intgen once to build interfaces to window manager routines, and again to build interfaces to a new data type. Both interfaces can be linked into Xlisp.

To use the generated files, you must compile the .c files and link them with all of the standard Xlisp object files. In addition, you must edit the file localdefs.h to contain an #include for each *.defs.h file, and edit the file localptrs.h to include each *.ptrs.h file. For example, suppose you run Intgen to build soundint.c, fugueint.c, and tableint.c. You would then edit localdefs.h to contain the following:

```
#include "soundintdefs.h"
#include "fugueintdefs.h"
#include "tableintdefs.h"
```

and edit localptrs.h to contain:

```
#include "soundintptrs.h"
#include "fugueintptrs.h"
#include "tableintptrs.h"
```

These localdefs.h and localptrs.h files are in turn included by xlftab.c which is where Xlisp builds a table of SUBRs.

To summarize, building an interface requires just a few simple steps:

- Write C code to be called by Xlisp interface routines. This C code does the real work, and in most cases is completely independent of Xlisp.
- Add comments to .h files to tell Intgen which routines to build interfaces to, and to specify the types of the arguments.
- Run Intgen to build interface routines.
- Edit localptrs.h and localdefs.h to include generated .h files.
- Compile and link Xlisp, including the new C code.

### B.2 Header file format

Each routine to be interfaced with Xlisp must be declared as follows:

```c
    type-name routine-name(); /* LISP: (func-name type1 type2 ...) */
```
The comment may be on the line following the declaration, but the declaration and the comment must each be on no more than one line. The characters LISP: at the beginning of the comment mark routines to put in the interface. The comment also gives the type and number of arguments. The function, when accessed from lisp will be known as \textit{func-name}, which need not bear any relationship to \textit{routine-name}. By convention, underscores in the C \textit{routine-name} should be converted to dashes in \textit{func-name}, and \textit{func-name} should be in all capitals. None of this is enforced or automated though.

Legal type \textit{names} are:

- \texttt{LVAL}
  - returns an Xlisp datum.
- \texttt{atom\_type}
  - equivalent to \texttt{LVAL}, but the result is expected to be an atom.
- \texttt{value\_type}
  - a value as used in Dannenberg’s score editor.
- \texttt{event\_type}
  - an event as used in Dannenberg’s score editor.
- \texttt{int}
  - interface will convert int to Xlisp \texttt{FIXNUM}.
- \texttt{boolean}
  - interface will convert int to \texttt{T} or \texttt{nil}.
- \texttt{float} or \texttt{double}
  - interface converts to \texttt{FLONUM}.
- \texttt{char *} or \texttt{string} or \texttt{string\_type}
  - interface converts to \texttt{STRING}. The result string will be copied into the XLISP heap.
- \texttt{void}
  - interface will return \texttt{nil}.

It is easy to extend this list. Any unrecognized type will be coerced to an int and then returned as a \texttt{FIXNUM}, and a warning will be issued.

The "*" after char must be followed by \textit{routine-name} with no intervening space.

Parameter types may be any of the following:

- \texttt{FIXNUM}
  - C routine expects an int.
- \texttt{FLONUM} or \texttt{FLOAT}
  - C routine expects a \texttt{double}.
- \texttt{STRING}
  - C routine expects \texttt{char *}, the string is not copied.
- \texttt{VALUE}
  - C routine expects a \texttt{value\_type}. (Not applicable to Fugue.)
- \texttt{EVENT}
  - C routine expects an \texttt{event\_type}. (Not applicable to Fugue.)
ANY
C routine expects LVAL.

ATOM
C routine expects LVAL which is a lisp atom.

FILE
C routine expects FILE *.

SOUND
C routine expects a SoundPtr.

Any of these may be followed by "*": FIXNUM*, FLONUM*, STRING*, ANY*, FILE*, indicating C routine expects int *, double *, char **, LVAL *, or FILE **. This is basically a mechanism for returning more than one value, not a mechanism for clobbering XLisp values. In this spirit, the interface copies the value (an int, double, char *, LVAL, or FILE *) to a local variable and passes the address of that variable to the C routine. On return, a list of resulting "*" parameters is constructed and bound to the global XLisp symbol *RSLT*. (Strings are copied.) If the C routine is void, then the result list is also returned by the corresponding XLisp function.

Note 1: this does not support C routines like strcpy that modify strings, because the C routine gets a pointer to the string in the XLisp heap. However, you can always add an intermediate routine that allocates space and then calls strcpy, or whatever.

Note 2: it follows that a new XLisp STRING will be created for each STRING* parameter.

Note 3: putting results on a (global!) symbol seems a bit unstructured, but note that one could write a multiple-value binding macro that hides this ugliness from the user if desired. In practice, I find that pulling the extra result values from *RSLT* when needed is perfectly acceptable.

For parameters that are result values only, the character "^" may be substituted for "*". In this case, the parameter is not to be passed in the XLisp calling site. However, the address of an initialized local variable of the given type is passed to the corresponding C function, and the resulting value is passed back through *RSLT* as ordinary result parameter as described above. The local variables are initialized to zero or NULL.

B.3 Using #define'd macros

If a comment of the form:

    /* LISP: type-name (routine-name-2 type-1 type-2 ...) */

appears on a line by itself and there was a #define on the previous line, then the preceding #define is treated as a C routine, e.g.

    #define leftshift(val, count) ((val) << (count))
    /* LISP: int (LOGSHIFT INT INT) */

will implement the LeLisp function LOGSHIFT.

The type-name following "LISP:" should have no spaces, e.g. use ANY*, not ANY *.
B.4 Lisp Include Files

Include files often define constants that we would like to have around in the Lisp world, but which are easier to initialize just by loading a text file. Therefore, a comment of the form:

```lisp
/* LISP-SRC: (any lisp expression) */
```

will cause Intgen to open a file `name.lsp` and append

(any lisp expression)

to `name.lsp`, where `name` is the interface name passed on the command line. If none of the include files examined have comments of this form, then no `name.lsp` file is generated. **Note:** the LISP-SRC comment must be on a new line.

B.5 Example

This file was used for testing Intgen. It uses a trick (ok, it’s a hack) to interface to a standard library macro (tolower). Since tolower is already defined, the macro ToLower is defined just to give Intgen a name to call. Two other routines, strlen and tough, are interfaced as well.

```lisp
/* igtest.h -- test interface for intgen */

#define ToLower(c) tolower(c)

/* LISP: int (TOLOWER FIXNUM) */

int strlen(); /* LISP: (STRLEN STRING) */

void tough();
   /* LISP: (TOUGH FIXNUM* FLONUM* STRING ANY FIXNUM) */
```

B.6 More Details

Intgen has some compiler switches to enable/disable the use of certain types, including VALUE and EVENT types used by Dannenberg’s score editing work, the SOUND type used by Fugue, and DEXT and SEXT types added for Dale Amon. Enabling all of these is not likely to cause problems, and the chances of an accidental use of these types getting through the compiler and linker seems very small.
Appendix C  XLISP: An Object-oriented Lisp

Version 2.0
February 6, 1988
by
David Michael Betz
127 Taylor Road
Peterborough, NH 03458
Copyright (c) 1988, by David Michael Betz
All Rights Reserved
Permission is granted for unrestricted non-commercial use

C.1  Introduction

XLISP is an experimental programming language combining some of the features of Common Lisp with an object-oriented extension capability. It was implemented to allow experimentation with object-oriented programming on small computers.

Implementations of XLISP run on virtually every operating system. XLISP is completely written in the programming language C and is easily extended with user written built-in functions and classes. It is available in source form to non-commercial users.

Many Common Lisp functions are built into XLISP. In addition, XLISP defines the objects Object and Class as primitives. Object is the only class that has no superclass and hence is the root of the class hierarchy tree. Class is the class of which all classes are instances (it is the only object that is an instance of itself).

This document is a brief description of XLISP. It assumes some knowledge of LISP and some understanding of the concepts of object-oriented programming.

I recommend the book *Lisp* by Winston and Horn and published by Addison Wesley for learning Lisp. The first edition of this book is based on MacLisp and the second edition is based on Common Lisp.

You will probably also need a copy of *Common Lisp: The Language* by Guy L. Steele, Jr., published by Digital Press to use as a reference for some of the Common Lisp functions that are described only briefly in this document.
C.2 A Note From The Author

If you have any problems with XLISP, feel free to contact me [me being David Betz - RBD] for help or advice. Please remember that since XLISP is available in source form in a high level language, many users [e.g. that Dannenberg fellow - RBD] have been making versions available on a variety of machines. If you call to report a problem with a specific version, I may not be able to help you if that version runs on a machine to which I don’t have access. Please have the version number of the version that you are running readily accessible before calling me.

If you find a bug in XLISP, first try to fix the bug yourself using the source code provided. If you are successful in fixing the bug, send the bug report along with the fix to me. If you don’t have access to a C compiler or are unable to fix a bug, please send the bug report to me and I’ll try to fix it.

Any suggestions for improvements will be welcomed. Feel free to extend the language in whatever way suits your needs. However, PLEASE DO NOT RELEASE ENHANCED VERSIONS WITHOUT CHECKING WITH ME FIRST!! I would like to be the clearing house for new features added to XLISP. If you want to add features for your own personal use, go ahead. But, if you want to distribute your enhanced version, contact me first. Please remember that the goal of XLISP is to provide a language to learn and experiment with LISP and object-oriented programming on small computers. I don’t want it to get so big that it requires megabytes of memory to run.

C.3 XLISP Command Loop

When XLISP is started, it first tries to load the workspace xlisp.wks from the current directory. If that file doesn’t exist, XLISP builds an initial workspace, empty except for the built-in functions and symbols.

Then XLISP attempts to load init.lsp from the current directory. It then loads any files named as parameters on the command line (after appending .lsp to their names).

XLISP then issues the following prompt:

>  

This indicates that XLISP is waiting for an expression to be typed.

When a complete expression has been entered, XLISP attempts to evaluate that expression. If the expression evaluates successfully, XLISP prints the result and then returns to the initial prompt waiting for another expression to be typed.

C.4 Special Characters

When XLISP is running from a console, some control characters invoke operations:

- Backspace and Delete characters erase the previous character on the input line (if any).
- Control-U erases the entire input line.
- Control-C executes the TOP-LEVEL function.
- Control-G executes the CLEAN-UP function.
- Control-P executes the CONTINUE function.
• Control-B stops execution and enters the break command loop. Execution can be continued by typing Control-P or (CONTINUE).
• Control-E turns on character echoing (Linux and Mac OS X only).
• Control-F turns off character echoing (Linux and Mac OS X only).
• Control-T evaluates the INFO function.

C.5 Break Command Loop

When XLISP encounters an error while evaluating an expression, it attempts to handle the error in the following way:

If the symbol *breakenable* is true, the message corresponding to the error is printed. If the error is correctable, the correction message is printed.

If the symbol *tracenable* is true, a trace back is printed. The number of entries printed depends on the value of the symbol *tracelimit*. If this symbol is set to something other than a number, the entire trace back stack is printed.

XLISP then enters a read/eval/print loop to allow the user to examine the state of the interpreter in the context of the error. This loop differs from the normal top-level read/eval/print loop in that if the user invokes the function continue, XLISP will continue from a correctable error. If the user invokes the function clean-up, XLISP will abort the break loop and return to the top level or the next lower numbered break loop. When in a break loop, XLISP prefixes the break level to the normal prompt.

If the symbol *breakenable* is nil, XLISP looks for a surrounding errset function. If one is found, XLISP examines the value of the print flag. If this flag is true, the error message is printed. In any case, XLISP causes the errset function call to return nil.

If there is no surrounding errset function, XLISP prints the error message and returns to the top level.

C.6 Data Types

There are several different data types available to XLISP programmers.

• lists
• symbols
• strings
• integers
• characters
• floats
• objects
• arrays
• streams
• subrs (built-in functions)
• fsubrs (special forms)
• closures (user defined functions)

C.7 The Evaluator

The process of evaluation in XLISP:

• Strings, integers, characters, floats, objects, arrays, streams, subrs, fsubrs and closures evaluate to themselves.
• Symbols act as variables and are evaluated by retrieving the value associated with their current binding.
• Lists are evaluated by examining the first element of the list and then taking one of the following actions:
  – If it is a symbol, the functional binding of the symbol is retrieved.
  – If it is a lambda expression, a closure is constructed for the function described by the lambda expression.
  – If it is a subr, fsubr or closure, it stands for itself.
  – Any other value is an error.
Then, the value produced by the previous step is examined:
  – If it is a subr or closure, the remaining list elements are evaluated and the subr or closure is called with these evaluated expressions as arguments.
  – If it is an fsubr, the fsubr is called using the remaining list elements as arguments (unevaluated).
  – If it is a macro, the macro is expanded using the remaining list elements as arguments (unevaluated). The macro expansion is then evaluated in place of the original macro call.

C.8 Lexical Conventions

The following conventions must be followed when entering XLISP programs:

Comments in XLISP code begin with a semi-colon character and continue to the end of the line.

Symbol names in XLISP can consist of any sequence of non-blank printable characters except the following:

( ) , ' " ;

Uppercase and lowercase characters are not distinguished within symbol names. All lowercase characters are mapped to uppercase on input.

Integer literals consist of a sequence of digits optionally beginning with a + or -. The range of values an integer can represent is limited by the size of a C long on the machine on which XLISP is running.

Floating point literals consist of a sequence of digits optionally beginning with a + or - and including an embedded decimal point. The range of values a floating point number can represent is limited by the size of a C float (double on machines with 32 bit addresses) on the machine on which XLISP is running.

Literal strings are sequences of characters surrounded by double quotes. Within quoted strings the "\" character is used to allow non-printable characters to be included. The codes recognized are:
C.9 Readtables

The behavior of the reader is controlled by a data structure called a readtable. The reader uses the symbol \*readtable\* to locate the current readtable. This table controls the interpretation of input characters. It is an array with 128 entries, one for each of the ASCII character codes. Each entry contains one of the following things:

- \*NIL\* – Indicating an invalid character
- \*:CONSTITUENT\* – Indicating a symbol constituent
- \*:WHITE-SPACE\* – Indicating a whitespace character
- \*(:TMACRO . fun)\* – Terminating readmacro
- \*(:NMACRO . fun)\* – Non-terminating readmacro
- \*:SESCAPE\* – Single escape character (\'\'
- \*:MESCAPE\* – Multiple escape character (\'|\'

In the case of \*:TMACRO\* and \*:NMACRO\*, the fun component is a function. This can either be a built-in readmacro function or a lambda expression. The function should take two parameters. The first is the input stream and the second is the character that caused the invocation of the readmacro. The readmacro function should return \*NIL\* to indicate that the character should be treated as white space or a value consed with \*NIL\* to indicate that the readmacro should be treated as an occurrence of the specified value. Of course, the readmacro code is free to read additional characters from the input stream.

XLISP defines several useful read macros:

- \'expr\* == (quote expr)*
- \#\'expr\* == (function expr)*
- \#(expr...\* == an array of the specified expressions
- \#x<hdigits>\* == a hexadecimal number (0-9,A-F)
- \#o<odigits>\* == an octal number (0-7)
- \#b<bdigits>\* == a binary number (0-1)
- \#\<char>\* == the ASCII code of the character
- \#| ... \# == a comment
C.10 Lambda Lists

There are several forms in XLISP that require that a "lambda list" be specified. A lambda list is a definition of the arguments accepted by a function. There are four different types of arguments.

The lambda list starts with required arguments. Required arguments must be specified in every call to the function.

The required arguments are followed by the &optional arguments. Optional arguments may be provided or omitted in a call. An initialization expression may be specified to provide a default value for an &optional argument if it is omitted from a call. If no initialization expression is specified, an omitted argument is initialized to NIL. It is also possible to provide the name of a supplied-p variable that can be used to determine if a call provided a value for the argument or if the initialization expression was used. If specified, the supplied-p variable will be bound to T if a value was specified in the call and NIL if the default value was used.

The &optional arguments are followed by the &rest argument. The &rest argument gets bound to the remainder of the argument list after the required and &optional arguments have been removed.

The &rest argument is followed by the &key arguments. When a keyword argument is passed to a function, a pair of values appears in the argument list. The first expression in the pair should evaluate to a keyword symbol (a symbol that begins with a ":"). The value of the second expression is the value of the keyword argument. Like &optional arguments, &key arguments can have initialization expressions and supplied-p variables. In addition, it is possible to specify the keyword to be used in a function call. If no keyword is specified, the keyword obtained by adding a ":" to the beginning of the keyword argument symbol is used. In other words, if the keyword argument symbol is foo, the keyword will be :foo.

The &key arguments are followed by the &aux variables. These are local variables that are bound during the evaluation of the function body. It is possible to have initialization expressions for the &aux variables.

Here is the complete syntax for lambda lists:
Appendix C. XLISP: An Object-oriented Lisp

C.11 Objects

Definitions:

- selector – a symbol used to select an appropriate method
- message – a selector and a list of actual arguments
- method – the code that implements a message

Since XLISP was created to provide a simple basis for experimenting with object-oriented programming, one of the primitive data types included is object. In XLISP, an object consists of a data structure containing a pointer to the object’s class as well as an array containing the values of the object’s instance variables.

Officially, there is no way to see inside an object (look at the values of its instance variables). The only way to communicate with an object is by sending it a message.

You can send a message to an object using the send function. This function takes the object as its first argument, the message selector as its second argument (which must be a symbol) and the message arguments as its remaining arguments.

The send function determines the class of the receiving object and attempts to find a method corresponding to the message selector in the set of messages defined for that class. If the message is not found in the object’s class and the class has a super-class, the search continues by looking at the messages defined for the super-class. This process continues from one super-class to the next until a method for the message is found. If no method is found, an error occurs.

When a method is found, the evaluator binds the receiving object to the symbol self and evaluates the method using the remaining elements of the original list as arguments to the method. These arguments are always evaluated prior to...
being bound to their corresponding formal arguments. The result of evaluating the method becomes the result of the expression.

Within the body of a method, a message can be sent to the current object by calling the \texttt{(send \ self \ ...)} . The method lookup starts with the object's class regardless of the class containing the current method.

Sometimes it is desirable to invoke a general method in a superclass even when it is overridden by a more specific method in a subclass. This can be accomplished by calling \texttt{send-super} , which begins the method lookup in the superclass of the class defining the current method rather than in the class of the current object.

The \texttt{send-super} function takes a selector as its first argument (which must be a symbol) and the message arguments as its remaining arguments. Notice that \texttt{send-super} can only be sent from within a method, and the target of the message is always the current object (\texttt{self} ). \texttt{(send-super \ ...)} is similar to \texttt{(send \ self \ ...)} except that method lookup begins in the superclass of the class containing the current method rather than the class of the current object.

\section*{C.12 The "Object" Class}

\texttt{Object} – the top of the class hierarchy.

Messages:

\texttt{:show} – show an object's instance variables.

returns – the object

\texttt{:class} – return the class of an object

returns – the class of the object

\texttt{:isa class} – test if object inherits from class

returns – \texttt{t} if object is an instance of \texttt{class} or a subclass of \texttt{class}, otherwise \texttt{nil}

\texttt{:isnew} – the default object initialization routine

returns – the object

\section*{C.13 The "Class" Class}

\texttt{Class} – class of all object classes (including itself)

Messages:

\texttt{:new} – create a new instance of a class

returns – the new class object

\texttt{:isnew ivars [cvars [super]]} – initialize a new class

\texttt{ivars} – the list of instance variable symbols

\texttt{cvars} – the list of class variable symbols

\texttt{super} – the superclass (default is \texttt{object})

returns – the new class object
:answer msg fargs code – add a message to a class
  msg – the message symbol
  fargs – the formal argument list (lambda list)
  code – a list of executable expressions
returns – the object

When a new instance of a class is created by sending the message :new to an existing class, the message :isnew followed by whatever parameters were passed to the :new message is sent to the newly created object.

When a new class is created by sending the :new message to the object Class, an optional parameter may be specified indicating the superclass of the new class. If this parameter is omitted, the new class will be a subclass of Object. A class inherits all instance variables, class variables, and methods from its super-class.

C.14 Profiling

The Xlisp 2.0 release has been extended with a profiling facility, which counts how many times and where eval is executed. A separate count is maintained for each named function, closure, or macro, and a count indicates an eval in the immediately (lexically) enclosing named function, closure, or macro. Thus, the count gives an indication of the amount of time spent in a function, not counting nested function calls. The list of all functions executed is maintained on the global *profile* variable. These functions in turn have *profile* properties, which maintain the counts. The profile system merely increments counters and puts symbols on the *profile* list. It is up to the user to initialize data and gather results. Profiling is turned on or off with the profile function. Unfortunately, methods cannot be profiled with this facility.

C.15 Symbols

• self - the current object (within a method context)
• *obarray* - the object hash table
• *standard-input* - the standard input stream
• *standard-output* - the standard output stream
• *error-output* - the error output stream
• *trace-output* - the trace output stream
• *debug-io* - the debug i/o stream
• *breakenable* - flag controlling entering break loop on errors
• *tracelist* - list of names of functions to trace
• *tracenable* - enable trace back printout on errors
• *tracelimit* - number of levels of trace back information
• *evalhook* - user substitute for the evaluator function
• *applyhook* - (not yet implemented)
• *readtable* - the current readtable
• *unbound* - indicator for unbound symbols
• *gc-flag* - controls the printing of gc messages
• *gc-hook* - function to call after garbage collection
• *integer-format* - format for printing integers (“%d” or “%ld”)
• *float-format* - format for printing floats (“%g”)
• *print-case* - symbol output case (:upcase or :downcase)

There are several symbols maintained by the read/eval/print loop. The symbols +, ++, and +++ are bound to the most recent three input expressions. The symbols *, ** and *** are bound to the most recent three results. The symbol - is bound to the expression currently being evaluated. It becomes the value of + at the end of the evaluation.

### C.16 Evaluation Functions

**eval(expr)** [SAL]
(eval expr) [LISP] – evaluate an xlisp expression
  expr – the expression to be evaluated
  returns – the result of evaluating the expression

**apply(fun, args)** [SAL]
(apply fun args) [LISP] – apply a function to a list of arguments
  fun – the function to apply (or function symbol)
  args – the argument list
  returns – the result of applying the function to the arguments

**funcall(fun, arg ...)** [SAL]
(funcall fun arg ...) [LISP] – call a function with arguments
  fun – the function to call (or function symbol)
  arg – arguments to pass to the function
  returns – the result of calling the function with the arguments

**quote(expr)** [SAL]
(quote expr) [LISP] – return an expression unevaluated
  expr – the expression to be quoted (quoted)
  returns – expr unevaluated

**function(expr)** [SAL]
(function expr) [LISP] – get the functional interpretation
  expr – the symbol or lambda expression (quoted)
  returns – the functional interpretation

**backquote(expr)** [SAL]
(backquote expr) [LISP] – fill in a template
  expr – the template
  returns – a copy of the template with comma and comma-at expressions expanded
lambda(args, expr ...) [SAL]
(lambda args expr ... ) [LISP] – make a function closure
   args – formal argument list (lambda list) (quoted)
   expr – expressions of the function body
   returns – the function closure

get-lambda-expression(closure) [SAL]
(get-lambda-expression closure) [LISP] – get the lambda expression
   closure – the closure
   returns – the original lambda expression

macroexpand(form) [SAL]
(macroexpand form) [LISP] – recursively expand macro calls
   form – the form to expand
   returns – the macro expansion

C.17 Symbol Functions

(set sym expr) [LISP] – set the value of a symbol. Note that in SAL, the function can be accessed as #set.
   sym – the symbol being set
   expr – the new value
   returns – the new value

setq([sym, expr] ...) [SAL]
(setq [sym expr] ...) [LISP] – set the value of a symbol. Note that in SAL, the set command is normally used.
   sym – the symbol being set (quoted)
   expr – the new value
   returns – the new value

psetq([sym, expr] ...) [SAL]
(psetq [sym expr] ...) [LISP] – parallel version of setq
   sym – the symbol being set (quoted)
   expr – the new value
   returns – the new value

setf([place, expr] ...) [SAL]
(setf [place expr] ...) [LISP] – set the value of a field
   place – the field specifier (quoted):
      sym – set value of a symbol
         (car expr) – set car of a cons node
         (cdr expr) – set cdr of a cons node
         (nth n expr) – set nth car of a list
         (aref expr n) – set nth element of an array
         (get sym prop) – set value of a property
   returns – the new value
(symbol-value sym) – set value of a symbol
(symbol-function sym) – set functional value of a symbol
(symbol-plist sym) – set property list of a symbol

expr – the new value
returns – the new value

(defun sym fargs expr ...) [LISP] – define a function
(defmacro sym fargs expr ...) [LISP] – define a macro
sym – symbol being defined (quoted)
fargs – formal argument list (lambda list) (quoted)
expr – expressions constituting the body of the
function (quoted) returns – the function symbol

gensym([tag]) [SAL]
gensym [tag]) [LISP] – generate a symbol
tag – string or number
returns – the new symbol

intern(pname) [SAL]
(intern pname) [LISP] – make an interned symbol
pname – the symbol’s print name string
returns – the new symbol

make-symbol(pname) [SAL]
(make-symbol pname) [LISP] – make an un interned symbol
pname – the symbol’s print name string
returns – the new symbol

symbol-name(sym) [SAL]
(symbol-name sym) [LISP] – get the print name of a symbol
sym – the symbol
returns – the symbol’s print name

symbol-value(sym) [SAL]
(symbol-value sym) [LISP] – get the value of a symbol
sym – the symbol
returns – the symbol’s value

symbol-function(sym) [SAL]
(symbol-function sym) [LISP] – get the functional value of a symbol
sym – the symbol
returns – the symbol’s functional value

symbol-plist(sym) [SAL]
(symbol-plist sym) [LISP] – get the property list of a symbol
sym – the symbol
returns – the symbol’s property list

hash(sym, n) [SAL]
(hash sym n) [LISP] – compute the hash index for a symbol
sym – the symbol or string
n – the table size (integer)
returns – the hash index (integer)
C.18 Property List Functions

\texttt{get(sym, prop)} [SAL]
\texttt{(get sym prop)} [LISP] – get the value of a property
\texttt{sym} – the symbol
\texttt{prop} – the property symbol
returns – the property value or \texttt{nil}

\texttt{putprop(sym, val, prop)} [SAL]
\texttt{(putprop sym val prop)} [LISP] – put a property onto a property list
\texttt{sym} – the symbol
\texttt{val} – the property value
\texttt{prop} – the property symbol
returns – the property value

\texttt{remprop(sym, prop)} [SAL]
\texttt{(remprop sym prop)} [LISP] – remove a property
\texttt{sym} – the symbol
\texttt{prop} – the property symbol
returns – \texttt{nil}

C.19 Array Functions

\texttt{aref(array, n)} [SAL]
\texttt{(aref array n)} [LISP] – get the nth element of an array
\texttt{array} – the array
\texttt{n} – the array index (integer)
returns – the value of the array element

\texttt{make-array(size)} [SAL]
\texttt{(make-array size)} [LISP] – make a new array
\texttt{size} – the size of the new array (integer)
returns – the new array

\texttt{vector(expr \ldots)} [SAL]
\texttt{(vector expr \ldots)} [LISP] – make an initialized vector
\texttt{expr} – the vector elements
returns – the new vector

C.20 List Functions

\texttt{car(expr)} [SAL]
\texttt{(car expr)} [LISP] – return the car of a list node
\texttt{expr} – the list node
returns – the car of the list node
cdr(expr) [SAL]
(cdr expr) [LISP] – return the cdr of a list node
  expr – the list node
  returns – the cdr of the list node

cxxr(expr) [SAL]
(cxxr expr) [LISP] – all cxxr combinations

cxxrr(expr) [SAL]
(cxxrr expr) [LISP] – all cxxrr combinations

cxxrrr(expr) [SAL]
(cxxrrr expr) [LISP] – all cxxrrr combinations

first(expr) [SAL]
(first expr) [LISP] – a synonym for car

second(expr) [SAL]
(second expr) [LISP] – a synonym for cadr

third(expr) [SAL]
(third expr) [LISP] – a synonym for caddr

fourth(expr) [SAL]
(fourth expr) [LISP] – a synonym for cadddr

rest(expr) [SAL]
(rest expr) [LISP] – a synonym for cdr

cons(expr1, expr2) [SAL]
(cons expr1 expr2) [LISP] – construct a new list node
  expr1 – the car of the new list node
  expr2 – the cdr of the new list node
  returns – the new list node

list(expr ...) [SAL]
(list expr ...) [LISP] – create a list of values
  expr – expressions to be combined into a list
  returns – the new list

append(expr ...) [SAL]
(append expr ...) [LISP] – append lists
  expr – lists whose elements are to be appended
  returns – the new list

reverse(expr) [SAL]
(reverse expr) [LISP] – reverse a list
  expr – the list to reverse
  returns – a new list in the reverse order

last(list) [SAL]
(last list) [LISP] – return the last list node of a list
  list – the list
  returns – the last list node in the list

member(expr, list, test: test, test-not: test-not) [SAL]
(member expr list &key :test :test-not) [LISP] – find an expression in a list
  expr – the expression to find
  list – the list to search
  :test – the test function (defaults to eql)
assoc(expr, alist, test: test, test-not: test-not) [SAL]
assoc expr alist &key :test :test-not [LISP] – find an expression in an a-list
  expr – the expression to find
  alist – the association list
  :test – the test function (defaults to eql)
  :test-not – the test function (sense inverted)
  returns – the alist entry or nil

remove(expr, list, test: test, test-not: test-not) [SAL]
(remove expr list &key :test :test-not) [LISP] – remove elements from a list
  expr – the element to remove
  list – the list
  :test – the test function (defaults to eql)
  :test-not – the test function (sense inverted)
  returns – copy of list with matching expressions removed

remove-if(test, list) [SAL]
(remove-if test list) [LISP] – remove elements that pass test
  test – the test predicate
  list – the list
  returns – copy of list with matching elements removed

remove-if-not(test, list) [SAL]
(remove-if-not test list) [LISP] – remove elements that fail test
  test – the test predicate
  list – the list
  returns – copy of list with non-matching elements removed

length(expr) [SAL]
(length expr) [LISP] – find the length of a list, vector or string
  expr – the list, vector or string
  returns – the length of the list, vector or string

nth(n, list) [SAL]
nth n list) [LISP] – return the nth element of a list
  n – the number of the element to return (zero origin)
  list – the list
  returns – the nth element or nil if the list isn’t that long

nthcdr(n, list) [SAL]
nthcdr n list) [LISP] – return the nth cdr of a list
  n – the number of the element to return (zero origin)
  list – the list
  returns – the nth cdr or nil if the list isn’t that long

mapc(fcn, list1, list ...) [SAL]
(mapc fcn list1 list ...) [LISP] – apply function to successive cars
  fcn – the function or function name
  listn – a list for each argument of the function
  returns – the first list of arguments
mapcar\(\textit{fcn}, \textit{list}_1, \textit{list} \ldots\) [SAL]
(mapcar \textit{fcn} \textit{list}_1 \textit{list} \ldots) [LISP] – apply function to successive cars
\textit{fcn} – the function or function name
\textit{list}_n – a list for each argument of the function
returns – a list of the values returned

mapl\(\textit{fcn}, \textit{list}_1, \textit{list} \ldots\) [SAL]
(mapl \textit{fcn} \textit{list}_1 \textit{list} \ldots) [LISP] – apply function to successive cdrs
\textit{fcn} – the function or function name
\textit{list}_n – a list for each argument of the function
returns – the first list of arguments

maplist\(\textit{fcn}, \textit{list}_1, \textit{list} \ldots\) [SAL]
(maplist \textit{fcn} \textit{list}_1 \textit{list} \ldots) [LISP] – apply function to successive cdrs
\textit{fcn} – the function or function name
\textit{list}_n – a list for each argument of the function
returns – a list of the values returned

subst\textit{(to, from, expr, test: test, test-not: test-not)} [SAL]
(subst \textit{to} \textit{from} \textit{expr} \textit{&key :test :test-not}) [LISP] – substitute expressions
\textit{to} – the new expression
\textit{from} – the old expression
\textit{expr} – the expression in which to do the substitutions
:test – the test function (defaults to eql)
:test-not – the test function (sense inverted)
returns – the expression with substitutions

sublis\textit{(alist, expr, test: test, test-not: test-not)} [SAL]
(sublis \textit{alist} \textit{expr} \textit{&key :test :test-not}) [LISP] – substitute with an a-list
\textit{alist} – the association list
\textit{expr} – the expression in which to do the substitutions
:test – the test function (defaults to eql)
:test-not – the test function (sense inverted)
returns – the expression with substitutions

C.21 Destructive List Functions

rplaca\(\textit{list}, \textit{expr}\) [SAL]
(rplaca \textit{list} \textit{expr}) [LISP] – replace the car of a list node
\textit{list} – the list node
\textit{expr} – the new value for the car of the list node
returns – the list node after updating the car

rplacd\(\textit{list}, \textit{expr}\) [SAL]
(rplacd \textit{list} \textit{expr}) [LISP] – replace the cdr of a list node
\textit{list} – the list node
\textit{expr} – the new value for the cdr of the list node
returns – the list node after updating the cdr
nconc(list ...) [SAL]
(nconc list ...) [LISP] – destructively concatenate lists

list – lists to concatenate
returns – the result of concatenating the lists

delete(expr, test: test, test-not: test-not) [SAL]
(delete expr &key :test :test-not) [LISP] – delete elements from a list

expr – the element to delete
list – the list
:test – the test function (defaults to eql)
:test-not – the test function (sense inverted)
returns – the list with the matching expressions deleted

delete-if(test, list) [SAL]
(delete-if test list) [LISP] – delete elements that pass test

test – the test predicate
list – the list
returns – the list with matching elements deleted

delete-if-not(test, list) [SAL]
(delete-if-not test list) [LISP] – delete elements that fail test

test – the test predicate
list – the list
returns – the list with non-matching elements deleted

sort(list, test) [SAL]
(sort list test) [LISP] – sort a list

list – the list to sort
:comparison – the comparison function
returns – the sorted list

Note: The comparison function should have two parameters and return true if the first parameter should come before the second parameter in the sorted result. For a list of numbers, built-in comparison functions can be used, e.g. (sort '(3 2 1) #'<) returns (1 2 3). To sort a list of lists by the first element of each list, you can write a function to access the keys and compare them, e.g. (sort '((3 a) (2 b) (1 c)) #'(lambda (x y) (< (car x) (car y)))) returns ((1 a) (2 b) (3 c)). In SAL, you could write function my-sort(a, b) return first(a) < first(b) print sort({{3 a} {2 b} {1 c}}, quote(my-sort)), which will print {{1 a} {2 b} {3 c}}.

C.22 Predicate Functions

atom(expr) [SAL]
(atom expr) [LISP] – is this an atom?

expr – the expression to check
returns – t if the value is an atom, nil otherwise

symbolp(expr) [SAL]
(symbolp expr) [LISP] – is this a symbol?

expr – the expression to check
returns – t if the expression is a symbol, nil otherwise
numberp(expr) [SAL]
(numberp expr) [LISP] – is this a number?
  expr – the expression to check
  returns – t if the expression is a number, nil otherwise

null(expr) [SAL]
(null expr) [LISP] – is this an empty list?
  expr – the list to check
  returns – t if the list is empty, nil otherwise

not(expr) [SAL]
(not expr) [LISP] – is this false?
  expr – the expression to check
  return – t if the value is nil, nil otherwise

listp(expr) [SAL]
(listp expr) [LISP] – is this a list?
  expr – the expression to check
  returns – t if the value is a cons or nil, nil otherwise

delp(list) [SAL]
(delp list) [LISP] – is this the end of a list
  list – the list
  returns – t if the value is nil, nil otherwise

consp(expr) [SAL]
(consp expr) [LISP] – is this a non-empty list?
  expr – the expression to check
  returns – t if the value is a cons, nil otherwise

integerp(expr) [SAL]
(integerp expr) [LISP] – is this an integer?
  expr – the expression to check
  returns – t if the value is an integer, nil otherwise

floatp(expr) [SAL]
(floatp expr) [LISP] – is this a float?
  expr – the expression to check
  returns – t if the value is a float, nil otherwise

stringp(expr) [SAL]
(stringp expr) [LISP] – is this a string?
  expr – the expression to check
  returns – t if the value is a string, nil otherwise

characterp(expr) [SAL]
(characterp expr) [LISP] – is this a character?
  expr – the expression to check
  returns – t if the value is a character, nil otherwise

arrayp(expr) [SAL]
(arrayp expr) [LISP] – is this an array?
  expr – the expression to check
  returns – t if the value is an array, nil otherwise
streamp(expr) [SAL]
(streamp expr) [LISP] – is this a stream?
    expr – the expression to check
    returns – t if the value is a stream, nil otherwise

objectp(expr) [SAL]
(objectp expr) [LISP] – is this an object?
    expr – the expression to check
    returns – t if the value is an object, nil otherwise

filep(expr) [SAL]
(filep expr) [LISP] – is this a file?
    expr – the expression to check
    returns – t if the value is an object, nil otherwise

boundp(sym) [SAL]
(boundp sym) [LISP] – is a value bound to this symbol?
    sym – the symbol
    returns – t if a value is bound to the symbol, nil otherwise

fboundp(sym) [SAL]
(fboundp sym) [LISP] – is a functional value bound to this symbol?
    sym – the symbol
    returns – t if a functional value is bound to the symbol, nil otherwise

minusp(expr) [SAL]
(minusp expr) [LISP] – is this number negative?
    expr – the number to test
    returns – t if the number is negative, nil otherwise

zerop(expr) [SAL]
(zerop expr) [LISP] – is this number zero?
    expr – the number to test
    returns – t if the number is zero, nil otherwise

plusp(expr) [SAL]
(plusp expr) [LISP] – is this number positive?
    expr – the number to test
    returns – t if the number is positive, nil otherwise

evenp(expr) [SAL]
(evenp expr) [LISP] – is this integer even?
    expr – the integer to test
    returns – t if the integer is even, nil otherwise

oddp(expr) [SAL]
(oddp expr) [LISP] – is this integer odd?
    expr – the integer to test
    returns – t if the integer is odd, nil otherwise

eq(expr1, expr2) [SAL]
(eq expr1 expr2) [LISP] – are the expressions identical?
    expr1 – the first expression
    expr2 – the second expression
    returns – t if they are equal, nil otherwise
C.23 Control Constructs

(cond (pair ...) [LISP] – evaluate conditionally
  pair – pair consisting of:
    (pred expr...)
  where:
    pred – is a predicate expression
    expr – evaluated if the predicate is not nil
  returns – the value of the first expression whose predicate is not nil

and(expr ...) [SAL]
(and expr ...) [LISP] – the logical and of a list of expressions
  expr – the expressions to be anded
  returns – nil if any expression evaluates to nil, otherwise the value of the last expression (evaluation of expressions stops after the first expression that evaluates to nil)

or(expr ...) [SAL]
(or expr ...) [LISP] – the logical or of a list of expressions
  expr – the expressions to be ored
  returns – nil if all expressions evaluate to nil, otherwise the value of the first non-nil expression (evaluation of expressions stops after the first expression that does not evaluate to nil)

(if texpr expr1 [expr2]) [LISP] – evaluate expressions conditionally. Note that the SAL conditional expression syntax is #?((test, iftrue-expression, iffalse-expression), but #if may be used instead of #?. Either form may omit the third argument, which defaults to nil.
  texpr – the test expression
    expr1 – the expression to be evaluated if texpr is non-nil
    expr2 – the expression to be evaluated if texpr is nil
  returns – the value of the selected expression.

when(texpr, expr ...) [SAL]
(when texpr expr ...) [LISP] – evaluate only when a condition is true
  texpr – the test expression
  expr – the expression(s) to be evaluated if texpr is non-nil
  returns – the value of the last expression or nil
unless(texpr, expr ...) [SAL]
(unless texpr expr ...) [LISP] – evaluate only when a condition is false
  texpr – the test expression
  expr – the expression(s) to be evaluated if texpr is nil
  returns – the value of the last expression or nil

(case expr case ...) [LISP] – select by case
  expr – the selection expression
  case – pair consisting of:
    (value expr...)
  where:
    value – is a single expression or a list of expressions (unevaluated)
    expr – are expressions to execute if the case matches
  returns – the value of the last expression of the matching case

(let (binding ...) expr ...) [LISP] – create local bindings
(let* (binding ...) expr ...) [LISP] – let with sequential binding
  binding – the variable bindings each of which is either:
    1) a symbol (which is initialized to nil)
    2) a list whose car is a symbol and whose cadr is an initialization expression
  expr – the expressions to be evaluated
  returns – the value of the last expression

(flet (binding ...) expr ...) [LISP] – create local functions
(labels (binding ...) expr ...) [LISP] – flet with recursive functions
(macrolet (binding ...) expr ...) [LISP] – create local macros
  binding – the function bindings each of which is:
    (sym fargs expr...)
  where:
    sym – the function/macro name
    fargs – formal argument list (lambda list)
    expr – expressions constituting the body of the function/macro
  expr – the expressions to be evaluated
  returns – the value of the last expression

catch(sym, expr ...) [SAL]
(catch sym expr ...) [LISP] – evaluate expressions and catch throws
  sym – the catch tag
  expr – expressions to evaluate
  returns – the value of the last expression the throw expression

throw(sym [, expr]) [SAL]
(throw sym [expr]) [LISP] – throw to a catch
  sym – the catch tag
  expr – the value for the catch to return (defaults to nil)
  returns – never returns

unwind-protect(expr, cexpr ...) [SAL]
(unwind-protect expr cexpr ...) [LISP] – protect evaluation of an expression
  expr – the expression to protect
  cexpr – the cleanup expressions
  returns – the value of the expression
Note: unwind-protect guarantees to execute the cleanup expressions even if a non-local exit terminates the evaluation of the protected expression

## C.24 Looping Constructs

(\texttt{loop expr ...}) \texttt{[LISP]} \textendash basic looping form
\begin{itemize}
  \item \textit{expr} \textendash the body of the loop
  \item \textit{returns} \textendash never returns (must use non-local exit)
\end{itemize}

(\texttt{do \{binding \ldots\} \{texpr rexpr \ldots\} \{expr \ldots\}}) \texttt{[LISP]}
(\texttt{do* \{binding \ldots\} \{texpr rexpr \ldots\} \{expr \ldots\}}) \texttt{[LISP]}
\begin{itemize}
  \item \textit{binding} \textendash the variable bindings each of which is either:
    \begin{enumerate}
      \item a symbol (which is initialized to \texttt{nil})
      \item a list of the form: \texttt{(sym init \{step\})} where:
        \begin{itemize}
          \item \textit{sym} \textendash is the symbol to bind
          \item \textit{init} \textendash is the initial value of the symbol
          \item \textit{step} \textendash is a step expression
        \end{itemize}
    \end{enumerate}
  \item \textit{texpr} \textendash the termination test expression
  \item \textit{rexpr} \textendash result expressions (the default is \texttt{nil})
  \item \textit{expr} \textendash the body of the loop (treated like an implicit \texttt{prog})
\end{itemize}
\begin{itemize}
  \item \textit{returns} \textendash the value of the last result expression
\end{itemize}

(\texttt{dolist \{sym expr [rexpr]\} \{expr \ldots\}}) \texttt{[LISP]} \textendash loop through a list
\begin{itemize}
  \item \textit{sym} \textendash the symbol to bind to each list element
  \item \textit{expr} \textendash the list expression
  \item \textit{rexpr} \textendash the result expression (the default is \texttt{nil})
  \item \textit{expr} \textendash the body of the loop (treated like an implicit \texttt{prog})
\end{itemize}

(\texttt{dotimes \{sym expr [rexpr]\} \{expr \ldots\}}) \texttt{[LISP]} \textendash loop from zero to n-1
\begin{itemize}
  \item \textit{sym} \textendash the symbol to bind to each value from 0 to n-1
  \item \textit{expr} \textendash the number of times to loop
  \item \textit{rexpr} \textendash the result expression (the default is \texttt{nil})
  \item \textit{expr} \textendash the body of the loop (treated like an implicit \texttt{prog})
\end{itemize}

## C.25 The Program Feature

(\texttt{prog \{binding \ldots\} \{expr \ldots\}}) \texttt{[LISP]} \textendash the program feature
(\texttt{prog* \{binding \ldots\} \{expr \ldots\}}) \texttt{[LISP]} \textendash prog with sequential binding
\begin{itemize}
  \item \textit{binding} \textendash the variable bindings each of which is either:
    \begin{enumerate}
      \item a symbol (which is initialized to \texttt{nil})
      \item a list whose car is a symbol and whose cadr is an initialization expression
    \end{enumerate}
  \item \textit{expr} \textendash expressions to evaluate or tags (symbols)
\end{itemize}
\begin{itemize}
  \item \textit{returns} \textendash \texttt{nil} or the argument passed to the return function
block(name, expr ...) [SAL]
(block name expr ...) [LISP] – named block
  name – the block name (symbol)
  expr – the block body
  returns – the value of the last expression

(return [expr]) [LISP] – cause a prog construct to return a value
  expr – the value (defaults to nil)
  returns – never returns

return-from(name [, value]) [SAL]
(return-from name [value]) [LISP] – return from a named block
  name – the block name (symbol)
  value – the value to return (defaults to nil)
  returns – never returns

tagbody(expr ...) [SAL]
(tagbody expr ...) [LISP] – block with labels
  expr – expression(s) to evaluate or tags (symbols)
  returns – nil

go(sym) [SAL]
(go sym) [LISP] – go to a tag within a tagbody or prog
  sym – the tag (quoted)
  returns – never returns

(progv slist vlist expr ...) [LISP] – dynamically bind symbols
  slist – list of symbols
  vlist – list of values to bind to the symbols
  expr – expression(s) to evaluate
  returns – the value of the last expression

prog1(expr1, expr ...) [SAL]
(prog1 expr1 expr ...) [LISP] – execute expressions sequentially
  expr1 – the first expression to evaluate
  expr – the remaining expressions to evaluate
  returns – the value of the first expression

prog2(expr1, expr2, expr ...) [SAL]
(prog2 expr1 expr2 expr ...) [LISP] – execute expressions sequentially
  expr1 – the first expression to evaluate
  expr2 – the second expression to evaluate
  expr – the remaining expressions to evaluate
  returns – the value of the second expression

progn(expr ...) [SAL]
(progn expr ...) [LISP] – execute expressions sequentially
  expr – the expressions to evaluate
  returns – the value of the last expression (or nil)
C.26 Debugging and Error Handling

trace(sym) [SAL]
(trace sym) [LISP] – add a function to the trace list
  sym – the function to add (quoted)
  returns – the trace list

untrace(sym) [SAL]
(untrace sym) [LISP] – remove a function from the trace list
  sym – the function to remove (quoted)
  returns – the trace list

error(emsg [, arg]) [SAL]
(error emsg [arg]) [LISP] – signal a non-correctable error
  emsg – the error message string
  arg – the argument expression (printed after the message)
  returns – never returns

cerror(cmsg, emsg [, arg]) [SAL]
(cerror cmsg emsg [arg]) [LISP] – signal a correctable error
  cmsg – the continue message string
  emsg – the error message string
  arg – the argument expression (printed after the message)
  returns – nil when continued from the break loop

break([bmsg [, arg]]) [SAL]
(break [bmsg [arg]]) [LISP] – enter a break loop
  bmsg – the break message string (defaults to **break**)
  arg – the argument expression (printed after the message)
  returns – nil when continued from the break loop

(clean-up) [LISP] – clean-up after an error
  returns – never returns

(top-level) [LISP] – clean-up after an error and return to the top level
  returns – never returns

(continue) [LISP] – continue from a correctable error
  returns – never returns

(errset expr [pflag]) [LISP] – trap errors
  expr – the expression to execute
  pflag – flag to control printing of the error message
  returns – the value of the last expression consed with nil
  or nil on error

(baktrace [n]) [LISP] – print n levels of trace back information
  n – the number of levels (defaults to all levels)
  returns – nil

(evalhook expr ehook ahook [env]) [LISP] – evaluate with hooks
  expr – the expression to evaluate
  ehook – the value for *evalhook*
ahook – the value for *applyhook*

env – the environment (default is nil)

returns – the result of evaluating the expression

\texttt{profile(\textit{flag})} [SAL]
\texttt{(profile \textit{flag})} [LISP] \textsuperscript{2} – turn profiling on or off.
\textit{flag} – nil turns profiling off, otherwise on

returns – the previous state of profiling.

\section*{C.27 Arithmetic Functions}

\texttt{truncate(\textit{expr})} [SAL]
\texttt{(truncate \textit{expr})} [LISP] – truncates a floating point number to an integer
\textit{expr} – the number

returns – the result of truncating the number

\texttt{float(\textit{expr})} [SAL]
\texttt{(float \textit{expr})} [LISP] – converts an integer to a floating point number
\textit{expr} – the number

returns – the result of floating the integer

\texttt{(\textit{+ expr} ...} [LISP] – add a list of numbers
\textit{expr} – the numbers

returns – the result of the addition

\texttt{(\textit{- expr} ...} [LISP] – subtract a list of numbers or negate a single number
\textit{expr} – the numbers

returns – the result of the subtraction

\texttt{(\textit{* expr} ...} [LISP] – multiply a list of numbers
\textit{expr} – the numbers

returns – the result of the multiplication

\texttt{(\textit{/ expr} ...} [LISP] – divide a list of numbers
\textit{expr} – the numbers

returns – the result of the division

\texttt{(1+ \textit{expr})} [LISP] – add one to a number
\textit{expr} – the number

returns – the number plus one

\texttt{(1- \textit{expr})} [LISP] – subtract one from a number
\textit{expr} – the number

returns – the number minus one

\texttt{rem(\textit{expr} ...} [SAL]
\texttt{(rem function \textit{expr} ...)} [LISP] – remainder of a list of numbers
\textit{expr} – the numbers

returns – the result of the remainder operation
\[\text{min}(expr \ldots) \] [SAL]
\(\text{min}(expr \ldots)\) [LISP] – the smallest of a list of numbers
\(expr\) – the expressions to be checked
returns – the smallest number in the list

\[\text{max}(expr \ldots) \] [SAL]
\(\text{max}(expr \ldots)\) [LISP] – the largest of a list of numbers
\(expr\) – the expressions to be checked
returns – the largest number in the list

\[\text{abs}(expr) \] [SAL]
\(\text{abs}(expr)\) [LISP] – the absolute value of a number
\(expr\) – the number
returns – the absolute value of the number

\[\text{gcd}(n1, n2 \ldots) \] [SAL]
\(\text{gcd}(n1 n2 \ldots)\) [LISP] – compute the greatest common divisor
\(n1\) – the first number (integer)
\(n2\) – the second number(s) (integer)
returns – the greatest common divisor

\[\text{random}(n) \] [SAL]
\(\text{random}(n)\) [LISP] – compute a random number between 0 and \(n\)-1 inclusive. If \(n\) is 0, return 0.
\(n\) – the upper bound (integer)
returns – a random number

\[\text{rrandom()} \] [SAL]
\(\text{rrandom()}\) [LISP] – compute a random real number between 0 and 1 inclusive
returns – a random floating point number

\[\text{random-seed}(n) \] [SAL]
\(\text{random-seed}(n)\) [LISP] – seed the random number generator with starting seed \(n\). If \text{random-seed} is not called, \text{sranddev} or some other initialization method will be used by default.
\(n\) – the upper bound (integer)
returns – a random number

\[\text{sin}(expr) \] [SAL]
\(\text{sin}(expr)\) [LISP] – compute the sine of a number
\(expr\) – the floating point number
returns – the sine of the number

\[\text{cos}(expr) \] [SAL]
\(\text{cos}(expr)\) [LISP] – compute the cosine of a number
\(expr\) – the floating point number
returns – the cosine of the number

\[\text{tan}(expr) \] [SAL]
\(\text{tan}(expr)\) [LISP] – compute the tangent of a number
\(expr\) – the floating point number
returns – the tangent of the number

\[\text{atan}(expr\ [\text{, expr2}]) \] [SAL]
\(\text{atan}(expr\ [\text{expr2}])\) [LISP] – compute the arctangent
\(expr\) – the value of \(x\)
\(expr2\) – the value of \(y\) (default value is 1.0)
returns – the arctangent of \(x/y\)
expt(x-expr, y-expr) [SAL]
(expt x-expr y-expr) [LISP] – compute x to the y power
  x-expr – the floating point number
  y-expr – the floating point exponent
  returns – x to the y power

exp(x-expr) [SAL]
(exp x-expr) [LISP] – compute e to the x power
  x-expr – the floating point number
  returns – e to the x power

sqrt(expr) [SAL]
(sqrt expr) [LISP] – compute the square root of a number
  expr – the floating point number
  returns – the square root of the number

(< n1 n2 ...) [LISP] – test for less than
(<= n1 n2 ...) [LISP] – test for less than or equal to
(= n1 n2 ...) [LISP] – test for equal to
(/= n1 n2 ...) [LISP] – test for not equal to
(> n1 n2 ...) [LISP] – test for greater than or equal to
(> n1 n2 ...) [LISP] – test for greater than
  n1 – the first number to compare
  n2 – the second number to compare
  returns – t if the results of comparing n1 with n2, n2 with n3, etc., are all true.

C.28 Bitwise Logical Functions

logand(expr ...) [SAL]
(logand expr ...) [LISP] – the bitwise and of a list of numbers
  expr – the numbers
  returns – the result of the and operation

logior(expr ...) [SAL]
(logior expr ...) [LISP] – the bitwise inclusive or of a list of numbers
  expr – the numbers
  returns – the result of the inclusive or operation

logxor(expr ...) [SAL]
(logxor expr ...) [LISP] – the bitwise exclusive or of a list of numbers
  expr – the numbers
  returns – the result of the exclusive or operation

lognot(expr) [SAL]
(lognot expr) [LISP] – the bitwise not of a number
  expr – the number
  returns – the bitwise inversion of number
C.29 String Functions

string(expr) [SAL]
(string expr) [LISP] – make a string from a value
  expr – an integer (which is first converted into its ASCII character value), string, character, or symbol
  returns – the string representation of the argument

string-search(pat, str, start: start, end: end) [SAL]
(string-search pat str &key :start :end) [LISP] – search for pattern in string
  pat – a string to search for
  str – the string to be searched
  :start – the starting offset in str
  :end – the ending offset + 1
  returns – index of pat in str or NIL if not found

string-trim(bag, str) [SAL]
(string-trim bag str) [LISP] – trim both ends of a string
  bag – a string containing characters to trim
  str – the string to trim
  returns – a trimmed copy of the string

string-left-trim(bag, str) [SAL]
(string-left-trim bag str) [LISP] – trim the left end of a string
  bag – a string containing characters to trim
  str – the string to trim
  returns – a trimmed copy of the string

string-right-trim(bag, str) [SAL]
(string-right-trim bag str) [LISP] – trim the right end of a string
  bag – a string containing characters to trim
  str – the string to trim
  returns – a trimmed copy of the string

string-upcase(str, start: start, end: end) [SAL]
(string-upcase str &key :start :end) [LISP] – convert to uppercase
  str – the string
  :start – the starting offset
  :end – the ending offset + 1
  returns – a converted copy of the string

string-downcase(str, start: start, end: end) [SAL]
(string-downcase str &key :start :end) [LISP] – convert to lowercase
  str – the string
  :start – the starting offset
  :end – the ending offset + 1
  returns – a converted copy of the string

nstring-upcase(str, start: start, end: end) [SAL]
(nstring-upcase str &key :start :end) [LISP] – convert to uppercase
  str – the string
  :start – the starting offset
  :end – the ending offset + 1
  returns – the converted string (not a copy)
nstring-downcase(str, start: start, end: end) [SAL]
(nstring-downcase str &key :start :end) [LISP] – convert to lowercase
   str – the string
   :start – the starting offset
   :end – the ending offset + 1
   returns – the converted string (not a copy)

strcat(expr ...) [SAL]
(strcat expr ...) [LISP] – concatenate strings
   expr – the strings to concatenate
   returns – the result of concatenating the strings

subseq(string, start [, end]) [SAL]
(subseq string start [end]) [LISP] – extract a substring
   string – the string
   start – the starting position (zero origin)
   end – the ending position + 1 (defaults to end)
   returns – substring between start and end

string<(str1, str2, start1: start1, end1: end1, start2: start2, end2: end2) [SAL]
(string< str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string<=(str1, str2, start1: start1, end1: end1, start2: start2, end2: end2) [SAL]
(string<= str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string==(str1, str2, start1: start1, end1: end1, start2: start2, end2: end2) [SAL]
(string= str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string/= (str1, str2, start1: start1, end1: end1, start2: start2, end2: end2) [SAL]
(string/= str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string>= (str1, str2, start1: start1, end1: end1, start2: start2, end2: end2) [SAL]
(string>= str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string> (str1, str2, start1: start1, end1: end1, start2: start2, end2: end2) [SAL]
(string> str1 str2 &key :start1 :end1 :start2 :end2) [LISP]

str1 – the first string to compare
   str2 – the second string to compare
   :start1 – first substring starting offset
   :end1 – first substring ending offset + 1
   :start2 – second substring starting offset
   :end2 – second substring ending offset + 1
   returns – t if predicate is true, nil otherwise
Note: case is significant with these comparison functions.
string-lessp(str1, str2, start1: start1, end1: end1,  
start2: start2, end2: end2) [SAL]
(string-lessp str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string-not-greaterp(str1, str2, start1: start1, end1: end1,  
start2: start2, end2: end2) [SAL]
(string-not-greaterp str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string-equal(str1, str2, start1: start1, end1: end1,  
start2: start2, end2: end2) [SAL]
(string-equal str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string-not-equal(str1, str2, start1: start1, end1: end1,  
start2: start2, end2: end2) [SAL]
(string-not-equal str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string-not-lessp(str1, str2, start1: start1, end1: end1,  
start2: start2, end2: end2) [SAL]
(string-not-lessp str1 str2 &key :start1 :end1 :start2 :end2) [LISP]
string-greaterp(str1, str2, start1: start1, end1: end1,  
start2: start2, end2: end2) [SAL]
(string-greaterp str1 str2 &key :start1 :end1 :start2 :end2) [LISP]

str1 – the first string to compare
str2 – the second string to compare
:start1 – first substring starting offset
:end1 – first substring ending offset + 1
:start2 – second substring starting offset
:end2 – second substring ending offset + 1
returns – t if predicate is true, nil otherwise
Note: case is not significant with these comparison functions.

C.30 Character Functions

char(string, index) [SAL]
(char string index) [LISP] – extract a character from a string
string – the string
index – the string index (zero relative)
returns – the ascii code of the character

upper-case-p(chr) [SAL]
(upper-case-p chr) [LISP] – is this an upper case character?
chr – the character
returns – t if the character is upper case, nil otherwise

lower-case-p(chr) [SAL]
(lower-case-p chr) [LISP] – is this a lower case character?
chr – the character
returns – t if the character is lower case, nil otherwise

both-case-p(chr) [SAL]
(both-case-p chr) [LISP] – is this an alphabetic (either case) character?
chr – the character
returns t if the character is alphabetic, nil otherwise

\begin{verbatim}
digit-char-p(chr) [SAL]
(digit-char-p chr) [LISP] – is this a digit character?
chr – the character
returns the digit weight if character is a digit, nil otherwise

char-code(chr) [SAL]
(char-code chr) [LISP] – get the ascii code of a character
chr – the character
returns the ascii character code (integer)

code-char(code) [SAL]
(code-char code) [LISP] – get the character with a specified ascii code
  code – the ascii code (integer)
  returns the character with that code or nil

char-upcase(chr) [SAL]
(char-upcase chr) [LISP] – convert a character to upper case
  chr – the character
  returns the upper case character

char-downcase(chr) [SAL]
(char-downcase chr) [LISP] – convert a character to lower case
  chr – the character
  returns the lower case character

digit-char(n) [SAL]
(digit-char n) [LISP] – convert a digit weight to a digit
  n – the digit weight (integer)
  returns the digit character or nil

char-int(chr) [SAL]
(char-int chr) [LISP] – convert a character to an integer
  chr – the character
  returns the ascii character code

int-char(int) [SAL]
(int-char int) [LISP] – convert an integer to a character
  int – the ascii character code
  returns the character with that code

char< (chr1, chr2 ...) [SAL]
(char< chr1 chr2 ...) [LISP]

char<= (chr1, chr2 ...) [SAL]
(char<= chr1 chr2 ...) [LISP]

char= (chr1, chr2 ...) [SAL]
(char= chr1 chr2 ...) [LISP]

char/= (chr1, chr2 ...) [SAL]
(char/= chr1 chr2 ...) [LISP]

char> (chr1, chr2 ...) [SAL]
(char> chr1 chr2 ...) [LISP]

char>=(chr1, chr2 ...) [SAL]
(char>=(chr1, chr2 ...) [LISP]

char>(chr1, chr2 ...) [SAL]
(char>(chr1, chr2 ...) [LISP]
\end{verbatim}
returns – t if predicate is true, nil otherwise
Note: case is significant with these comparison functions.

\texttt{char-lessp(chr1, chr2 ...)} [SAL]
\texttt{(char-lessp chr1 chr2 ...)} [LISP]
\texttt{char-not-greaterp(chr1, chr2 ...)} [SAL]
\texttt{(char-not-greaterp chr1 chr2 ...)} [LISP]
\texttt{char-equal(chr1, chr2 ...)} [SAL]
\texttt{(char-equal chr1 chr2 ...)} [LISP]
\texttt{char-not-equal(chr1, chr2 ...)} [SAL]
\texttt{(char-not-equal chr1 chr2 ...)} [LISP]
\texttt{char-not-lessp(chr1, chr2 ...)} [SAL]
\texttt{(char-not-lessp chr1 chr2 ...)} [LISP]
\texttt{char-greaterp(chr1, chr2 ...)} [SAL]
\texttt{(char-greaterp chr1 chr2 ...)} [LISP]

\textit{chr1} – the first string to compare
\textit{chr2} – the second string(s) to compare
returns – t if predicate is true, nil otherwise
Note: case is not significant with these comparison functions.

### C.31 Input/Output Functions

\texttt{read([stream [, eof [, rflag]]])} [SAL]
\texttt{(read [stream [eof [rflag]]])} [LISP] – read an expression
\textit{stream} – the input stream (default is standard input)
\textit{eof} – the value to return on end of file (default is \texttt{nil})
\textit{rflag} – recursive read flag (default is \texttt{nil})
returns – the expression read

\texttt{(print expr [stream])} [LISP] – print an expression on a new line
\textit{expr} – the expression to be printed
\textit{stream} – the output stream (default is standard output)
returns – the expression

\texttt{prin1(expr [, stream])} [SAL]
\texttt{(prin1 expr [stream])} [LISP] – print an expression
\textit{expr} – the expression to be printed
\textit{stream} – the output stream (default is standard output)
returns – the expression

\texttt{princ(expr [, stream])} [SAL]
\texttt{(princ expr [stream])} [LISP] – print an expression without quoting
\textit{expr} – the expressions to be printed
\textit{stream} – the output stream (default is standard output)
returns – the expression

\texttt{pprint(expr [, stream])} [SAL]
\texttt{(pprint expr [stream])} [LISP] – pretty print an expression
\textit{expr} – the expressions to be printed
stream – the output stream (default is standard output)
returns – the expression

\( \text{terpri}([\text{stream}]) \) [SAL]
\( \text{terpri}([\text{stream}]) \) [LISP] – terminate the current print line
stream – the output stream (default is standard output)
returns – nil

flatsize(\( \text{expr} \)) [SAL]
(flatsize \( \text{expr} \)) [LISP] – length of printed representation using prin1
expr – the expression
returns – the length

flatc(\( \text{expr} \)) [SAL]
(flatc \( \text{expr} \)) [LISP] – length of printed representation using princ
expr – the expression
returns – the length

### C.32 The Format Function

\( \text{format}([\text{stream, fmt, arg} \ldots]) \) [SAL]
\( \text{format}([\text{stream, fmt, arg} \ldots]) \) [LISP] – do formated output
stream – the output stream
fmt – the format string
arg – the format arguments
returns – output string if \( \text{stream} \) is nil, nil otherwise

The format string can contain characters that should be copied directly to the output and formatting directives. The formatting directives are:

\~\text{A} – print next argument using princ
\~\text{S} – print next argument using prin1
\~\text{\%} – start a new line
\~\text{\textasciitilde} – print a tilde character
\~\text{<newline>} – ignore this one newline and white space on the next line up to the first non-white-space character or newline. This allows strings to continue across multiple lines

### C.33 File I/O Functions

Note that files are ordinarily opened as text. Binary files (such as standard midi files) must be opened with \texttt{open-binary} on non-unix systems.
open\( (\)fname, direction: direction \) [SAL]  
\( (open \) \)fname \&key :direction) [LISP] – open a file stream  
\( fname \) – the file name string or symbol  
\( :direction \) – :input or :output (default is :input)  
returns – a stream

open-binary\( (\)fname, direction: direction \) [SAL]  
\( (open-binary \) \)fname \&key :direction) [LISP] – open a binary file stream  
\( fname \) – the file name string or symbol  
\( :direction \) – :input or :output (default is :input)  
returns – a stream

close\( (\)stream \) [SAL]  
\( (close \) \)stream \) [LISP] – close a file stream  
\( stream \) – the stream  
returns – nil

setdir\( (\)path [, verbose]) [SAL]  
\( (setdir \) \)path [verbose]) [LISP] – set current directory  
\( path \) – the path of the new directory  
\( verbose \) – print error message if current directory cannot be changed to path  
returns – the resulting full path, e.g. \( \) (setdir ".\”) gets the current working directory, or nil if an error occurs

listdir\( (\)path \) [SAL]  
\( (listdir \) \)path \) [LISP] – get a directory listing  
\( path \) – the path of the directory to be listed  
returns – list of filenames in the directory

get-temp-path\() [SAL]  
\( (get-temp-path \) \) [LISP] – get a path where a temporary file can be created. Under Windows, this is based on environment variables. If XLISP is running as a sub-process to Java, the environment may not exist, in which case the default result is the unfortunate choice \( c:\windows\).  
returns – the resulting full path as a string

get-user\() [SAL]  
\( (get-user \) \) [LISP] – get the user ID. In Unix systems (including OS X and Linux), this is the value of the USER environment variable. In Windows, this is currently just "nyquist", which is also returned if the environment variable cannot be accessed. This function is used to avoid the case of two users creating files of the same name in the same temp directory.  
returns – the string naming the user

find-in-xlisp-path\( (\)filename \) [SAL]  
\( (find-in-xlisp-path \) \)filename \) [LISP] – search the XLISP search path (e.g. XLISPPATH from the environment) for filename. If filename is not found as is, and there is no file extension, append ".lsp" to filename and search again. The current directory is not searched.  
\( filename \) – the name of the file to search for  
returns – a full path name to the first occurrence found

read-char\( (\)[stream]) [SAL]  
\( (read-char \) \)[stream]) [LISP] – read a character from a stream  
\( stream \) – the input stream (default is standard input)  
returns – the character

peek-char\( (\)[flag [, stream]]) [SAL]  
\( (peek-char \) \)[flag [stream]]) [LISP] – peek at the next character  
\( flag \) – flag for skipping white space (default is nil)
stream – the input stream (default is standard input)
returns – the character (integer)

(write-char ch [stream]) [SAL]
(write-char ch [stream]) [LISP] – write a character to a stream
ch – the character to write
stream – the output stream (default is standard output)
returns – the character

(read-int [stream [length]]) [SAL]
(read-int [stream [length]]) [LISP] – read a binary integer from a stream
stream – the input stream (default is standard input)
length – the length of the integer in bytes (default is 4)
returns – the integer
Note: Integers are assumed to be big-endian (high-order byte first) and signed, regardless of the platform. To read little-endian format, use a negative number for the length, e.g. -4 indicates a 4-bytes, low-order byte first. The file should be opened in binary mode.

(write-int ch [stream [length]]) [SAL]
(write-int ch [stream [length]]) [LISP] – write a binary integer to a stream
ch – the character to write
stream – the output stream (default is standard output)
length – the length of the integer in bytes (default is 4)
returns – the integer
Note: Integers are assumed to be big-endian (high-order byte first) and signed, regardless of the platform. To write in little-endian format, use a negative number for the length, e.g. -4 indicates a 4-bytes, low-order byte first. The file should be opened in binary mode.

(read-float [stream [length]]) [SAL]
(read-float [stream [length]]) [LISP] – read a binary floating-point number from a stream
stream – the input stream (default is standard input)
length – the length of the float in bytes (default is 4, legal values are -4, -8, 4, and 8)
returns – the integer
Note: Floats are assumed to be big-endian (high-order byte first) and signed, regardless of the platform. To read little-endian format, use a negative number for the length, e.g. -4 indicates a 4-bytes, low-order byte first. The file should be opened in binary mode.

(write-float ch [stream [length]]) [SAL]
(write-float ch [stream [length]]) [LISP] – write a binary floating-point number to a stream
ch – the character to write
stream – the output stream (default is standard output)
length – the length of the float in bytes (default is 4, legal values are -4, -8, 4, and 8)
returns – the integer
Note: Floats are assumed to be big-endian (high-order byte first) and signed, regardless of the platform. To write in little-endian format, use a negative number for the length, e.g. -4 indicates a 4-bytes, low-order byte first. The file should be opened in binary mode.

(read-line [stream]) [SAL]
(read-line [stream]) [LISP] – read a line from a stream
stream – the input stream (default is standard input)
returns – the string
read-byte([stream]) [SAL]
(read-byte [stream]) [LISP] – read a byte from a stream
  stream – the input stream (default is standard input)
  returns – the byte (integer)

write-byte(byte [, stream]) [SAL]
(write-byte byte [stream]) [LISP] – write a byte to a stream
  byte – the byte to write (integer)
  stream – the output stream (default is standard output)
  returns – the byte (integer)

## C.34 String Stream Functions

These functions operate on unnamed streams. An unnamed output stream collects characters sent to it when it is used as
the destination of any output function. The functions get-output-stream-string and get-output-stream-list
return a string or a list of characters.

An unnamed input stream is setup with the make-string-input-stream function and returns each character of the
string when it is used as the source of any input function.

make-string-input-stream(str [, start [, end]]) [SAL]
(make-string-input-stream str [start [end]]) [LISP]
  str – the string
  start – the starting offset
  end – the ending offset + 1
  returns – an unnamed stream that reads from the string

make-string-output-stream() [SAL]
(make-string-output-stream) [LISP]
  returns – an unnamed output stream

get-output-stream-string(stream) [SAL]
(get-output-stream-string stream) [LISP]
  stream – the output stream
  returns – the output so far as a string
  Note: the output stream is emptied by this function

get-output-stream-list(stream) [SAL]
(get-output-stream-list stream) [LISP]
  stream – the output stream
  returns – the output so far as a list
  Note: the output stream is emptied by this function
C.35 System Functions

Note: the load function first tries to load a file from the current directory. A .lsp extension is added if there is not already an alphanumeric extension following a period. If that fails, XLISP searches the path, which is obtained from the XLISPPATH environment variable in Unix and HKEY_LOCAL_MACHINE\SOFTWARE\CMU\Nyquist\XLISPPATH under Win32. (The Macintosh version has no search path.)

\begin{Verbatim}
\textbf{get-env(name)} \textit{[SAL]}

\textbf{(get-env name)} \textit{[LISP]} \textend{Verbatim} – get from an environment variable
\begin{itemize}
\item \textit{name} – the name of the environment variable
\item \textit{returns} – string value of the environment variable, nil if variable does not exist
\end{itemize}

\begin{Verbatim}
\textbf{(load fname &key :verbose :print)} \textit{[LISP]} \textend{Verbatim} – load a source file
\begin{itemize}
\item \textit{fname} – the filename string or symbol
\item \textit{:verbose} – the verbose flag (default is t)
\item \textit{:print} – the print flag (default is nil)
\item \textit{returns} – the filename
\end{itemize}

\begin{Verbatim}
\textbf{save(fname)} \textit{[SAL]}

\textbf{(save fname)} \textit{[LISP]} \textend{Verbatim} – save workspace to a file
\begin{itemize}
\item \textit{fname} – the filename string or symbol
\item \textit{returns} – t if workspace was written, nil otherwise
\end{itemize}

\begin{Verbatim}
\textbf{restore(fname)} \textit{[SAL]}

\textbf{(restore fname)} \textit{[LISP]} \textend{Verbatim} – restore workspace from a file
\begin{itemize}
\item \textit{fname} – the filename string or symbol
\item \textit{returns} – nil on failure, otherwise never returns
\end{itemize}

\begin{Verbatim}
\textbf{dribble([fname])} \textit{[SAL]}

\textbf{(dribble [fname])} \textit{[LISP]} \textend{Verbatim} – create a file with a transcript of a session
\begin{itemize}
\item \textit{fname} – file name string or symbol (if missing, close current transcript)
\item \textit{returns} – t if the transcript is opened, nil if it is closed
\end{itemize}

\begin{Verbatim}
\textbf{gc()} \textit{[SAL]}

\textbf{(gc)} \textit{[LISP]} \textend{Verbatim} – force garbage collection
\begin{itemize}
\item \textit{returns} – nil
\end{itemize}

\begin{Verbatim}
\textbf{expand(num)} \textit{[SAL]}

\textbf{(expand num)} \textit{[LISP]} \textend{Verbatim} – expand memory by adding segments
\begin{itemize}
\item \textit{num} – the number of segments to add
\item \textit{returns} – the number of segments added
\end{itemize}

\begin{Verbatim}
\textbf{alloc(num)} \textit{[SAL]}

\textbf{(alloc num)} \textit{[LISP]} \textend{Verbatim} – change number of nodes to allocate in each segment
\begin{itemize}
\item \textit{num} – the number of nodes to allocate
\item \textit{returns} – the old number of nodes to allocate
\end{itemize}

\begin{Verbatim}
\textbf{info()} \textit{[SAL]}

\textbf{(info)} \textit{[LISP]} \textend{Verbatim} – show information about memory usage.
\begin{itemize}
\item \textit{returns} – nil
\end{itemize}

\begin{Verbatim}
\textbf{room()} \textit{[SAL]}

\textbf{(room)} \textit{[LISP]} \textend{Verbatim} – show memory allocation statistics
\begin{itemize}
\item \textit{returns} – nil
\end{itemize}
type-of( expr ) [SAL]
(type-of expr) [LISP] – returns the type of the expression
   expr – the expression to return the type of
   returns – nil if the value is nil otherwise one of the symbols:
      SYMBOL – for symbols
      OBJECT – for objects
      CONS – for conses
      SUBR – for built-in functions
      FSUBR – for special forms
      CLOSURE – for defined functions
      STRING – for strings
      FIXNUM – for integers
      FLONUM – for floating point numbers
      CHARACTER – for characters
      FILE-STREAM – for file pointers
      UNNAMED-STREAM – for unnamed streams
      ARRAY – for arrays

peek( addrs ) [SAL]
(peek addrs) [LISP] – peek at a location in memory
   addrs – the address to peek at (integer)
   returns – the value at the specified address (integer)

poke( addrs , value ) [SAL]
(poke addrs value) [LISP] – poke a value into memory
   addrs – the address to poke (integer)
   value – the value to poke into the address (integer)
   returns – the value

bigendianp() [SAL]
(bigendianp) [LISP] – is this a big-endian machine?
   returns – T if this a big-endian architecture, storing the high-order byte of an integer at the lowest byte address of
   the integer; otherwise, NIL.  

address-of( expr ) [SAL]
(address-of expr) [LISP] – get the address of an xlisp node
   expr – the node
   returns – the address of the node (integer)

exit() [SAL]
(exit) [LISP] – exit xlisp. (Note: in Audacity plug-ins, exit is undefined because exiting would terminate
   Audacity.)
   returns – never returns

setup-console() [SAL]
(setup-console) [LISP] – set default console attributes
   returns – NIL
   Note: Under Windows, Nyquist normally starts up in a medium-sized console window with black text and a white
   background, with a window title of "Nyquist." This is normally accomplished by calling setup-console in
   system.lsp. In Nyquist, you can avoid this behavior by setting *setup-console* to NIL in your init.lsp
   file. If setup-console is not called, Nyquist uses standard input and output as is. This is what you want if you
   are running Nyquist inside of emacs, for example.

---

\[10\] This is not a standard XLISP 2.0 function.
echoenabled(flag) [SAL]

(\texttt{echoenabled} \textit{flag}) [LISP] – turn console input echoing on or off
\textit{flag} – T to enable echo, NIL to disable
returns – NIL

Note: This function is only implemented under Linux and Mac OS X. If Nyquist I/O is redirected through pipes, the Windows version does not echo the input, but the Linux and Mac versions do. You can turn off echoing with this function. Under windows it is defined to do nothing.

\section*{C.36 File I/O Functions}

\subsection*{C.36.1 Input from a File}

To open a file for input, use the \texttt{open} function with the keyword argument \texttt{:direction} set to \texttt{:input}. To open a file for output, use the \texttt{open} function with the keyword argument \texttt{:direction} set to \texttt{:output}. The \texttt{open} function takes a single required argument which is the name of the file to be opened. This name can be in the form of a string or a symbol. The \texttt{open} function returns an object of type \texttt{FILE-STREAM} if it succeeds in opening the specified file. It returns the value \texttt{nil} if it fails. In order to manipulate the file, it is necessary to save the value returned by the \texttt{open} function. This is usually done by assigning it to a variable with the \texttt{setq} special form or by binding it using \texttt{let} or \texttt{let*}. Here is an example:

\begin{verbatim}
(setq fp (open "init.lsp" :direction :input))
\end{verbatim}

Evaluating this expression will result in the file \texttt{init.lsp} being opened. The file object that will be returned by the \texttt{open} function will be assigned to the variable \texttt{fp}.

It is now possible to use the file for input. To read an expression from the file, just supply the value of the \texttt{fp} variable as the optional \texttt{stream} argument to \texttt{read}.

\begin{verbatim}
(read fp)
\end{verbatim}

Evaluating this expression will result in reading the first expression from the file \texttt{init.lsp}. The expression will be returned as the result of the \texttt{read} function. More expressions can be read from the file using further calls to the \texttt{read} function. When there are no more expressions to read, the \texttt{read} function will return \texttt{nil} (or whatever value was supplied as the second argument to \texttt{read}).

Once you are done reading from the file, you should close it. To close the file, use the following expression:

\begin{verbatim}
(close fp)
\end{verbatim}

Evaluating this expression will cause the file to be closed.

\subsection*{C.36.2 Output to a File}

Writing to a file is pretty much the same as reading from one. You need to open the file first. This time you should use the \texttt{open} function to indicate that you will do output to the file. For example:

\begin{verbatim}
(setq fp (open "test.dat" :direction :output))
\end{verbatim}
Evaluating this expression will open the file \texttt{test.dat} for output. If the file already exists, its current contents will be discarded. If it doesn’t already exist, it will be created. In any case, a FILE-STREAM object will be returned by the \texttt{OPEN} function. This file object will be assigned to the \texttt{fp} variable.

It is now possible to write to this file by supplying the value of the \texttt{fp} variable as the optional \texttt{stream} parameter in the \texttt{print} function.

\begin{verbatim}
(print "Hello there" fp)
\end{verbatim}

Evaluating this expression will result in the string "Hello there" being written to the file \texttt{test.dat}. More data can be written to the file using the same technique.

Once you are done writing to the file, you should close it. Closing an output file is just like closing an input file.

\begin{verbatim}
(close fp)
\end{verbatim}

Evaluating this expression will close the output file and make it permanent.

\subsection*{C.36.3 A Slightly More Complicated File Example}

This example shows how to open a file, read each Lisp expression from the file and print it. It demonstrates the use of files and the use of the optional \texttt{stream} argument to the \texttt{read} function.

\begin{verbatim}
(do* ((fp (open "test.dat" :direction :input))
     (ex (read fp) (read fp)))
     ((null ex) nil)
     (print ex))
\end{verbatim}
Index

*, 53, 247
*=, 62
*=tolerance*, 130
*a4-hertz*, 74, 130
*applyhook*, 231
*audio-markers*, 106
*autonorm*, 130
*autonorm-max-samples*, 130
*autonorm-previous-peak*, 130
*autonorm-target*, 130
*autonorm-type*, 131
*autonormflag*, 130
*breakenable*, 131, 225, 231
*clipping-error*, 131
*clipping-threshold*, 131
*control-srate*, 28, 102, 131
*debug-io*, 231
*default-control-srate*, 131
*default-plot-file*, 111, 131
*default-sf-bits*, 131
*default-sf-dir*, 106, 131
*default-sf-format*, 131
*default-sf-srate*, 109, 131
*default-sound-srate*, 132
*error-output*, 231
*evalhook*, 231
*file-separator*, 132
*float-format*, 232
*gc-flag*, 232
*gc-hook*, 232
*integer-format*, 232
*loud*, 28
*lpslider-cutoff*, 132
*obarray*, 231
*print-case*, 232
*readtable*, 227, 232
*rslt*, 132, 221
*saw-table*, 132
*sine-table*, 132
*snd-device*, 132
*snd-list-devices*, 132
*sound-srate*, 29, 102, 132
*soundenable*, 132
*spec-plot-bw*, 112
*spec-plot-db*, 112
*spec-plot-res*, 111
*standard-input*, 231
*standard-output*, 231
*start*, 28
*stop*, 28
*sustain*, 28
*table*, 130
*trace-output*, 231
*tracelimit*, 225, 231
*tracelist*, 231
*tracenable*, 132, 225, 231
*transpose*, 28
*tri-table*, 133
*unbound*, 232
*warp*, 28, 101
*, 53, 247
+=, 62
, (adagio), 152
-, 53, 247
/, 53, 247
/=, 249
:answer, 231
:class, 230
:isa, 230
:isnew, 230
:new, 230
:show, 230
:(adagio), 152
<, 53, 249
<=, 54, 63, 249
=, 53, 249
>, 53, 249

263
<table>
<thead>
<tr>
<th>Symbol</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>&gt;=, 53</td>
<td>53, 63, 249</td>
</tr>
<tr>
<td># (adagio articulation), 146</td>
<td></td>
</tr>
<tr>
<td>#?, sal, 55</td>
<td></td>
</tr>
<tr>
<td>#define'd macros, 221</td>
<td></td>
</tr>
<tr>
<td>#f, 53</td>
<td></td>
</tr>
<tr>
<td>#t, 53</td>
<td></td>
</tr>
<tr>
<td>%, 53</td>
<td></td>
</tr>
<tr>
<td>% (adagio thirtysecond note), 145</td>
<td></td>
</tr>
<tr>
<td>&amp;, 54</td>
<td></td>
</tr>
<tr>
<td>&amp;=?, 62</td>
<td></td>
</tr>
<tr>
<td>^, 53</td>
<td></td>
</tr>
<tr>
<td>^=, 62</td>
<td></td>
</tr>
<tr>
<td>^ (adagio sixtyfourth note), 145</td>
<td></td>
</tr>
<tr>
<td>~, 54</td>
<td></td>
</tr>
<tr>
<td>~=, 54</td>
<td></td>
</tr>
<tr>
<td>~ (adagio), 153</td>
<td></td>
</tr>
<tr>
<td>1+, 247</td>
<td></td>
</tr>
<tr>
<td>1-, 247</td>
<td></td>
</tr>
<tr>
<td>A440, 19, 74</td>
<td></td>
</tr>
<tr>
<td>Abs, 248</td>
<td></td>
</tr>
<tr>
<td>Abs-env, 101</td>
<td></td>
</tr>
<tr>
<td>Absolute stretch, sal, 54</td>
<td></td>
</tr>
<tr>
<td>Absolute time shift, sal, 54</td>
<td></td>
</tr>
<tr>
<td>Absolute value, 97, 114</td>
<td></td>
</tr>
<tr>
<td>Access samples, 69</td>
<td></td>
</tr>
<tr>
<td>Accidentals, 145</td>
<td></td>
</tr>
<tr>
<td>Accumulate pattern, 171</td>
<td></td>
</tr>
<tr>
<td>Accumulation pattern, 171</td>
<td></td>
</tr>
<tr>
<td>Adagio, 143</td>
<td></td>
</tr>
<tr>
<td>Adagio file readr, 142</td>
<td></td>
</tr>
<tr>
<td>Adagio file writer, 143</td>
<td></td>
</tr>
<tr>
<td>Add offset to sound, 115</td>
<td></td>
</tr>
<tr>
<td>Add to file samples, 110</td>
<td></td>
</tr>
<tr>
<td>Add-action-to-workspace, 192</td>
<td></td>
</tr>
<tr>
<td>Add-to-workspace, 191</td>
<td></td>
</tr>
<tr>
<td>Additive synthesis, gongs, 20</td>
<td></td>
</tr>
<tr>
<td>Address-of, 260</td>
<td></td>
</tr>
<tr>
<td>Aftertouch, 153</td>
<td></td>
</tr>
<tr>
<td>Age, 196</td>
<td></td>
</tr>
<tr>
<td>Algorithmic composition, 166</td>
<td></td>
</tr>
<tr>
<td>All pass filter, 87</td>
<td></td>
</tr>
<tr>
<td>Alloc, 259</td>
<td></td>
</tr>
<tr>
<td>Allpass2, 91</td>
<td></td>
</tr>
<tr>
<td>Allpoles-from-lpc, 160</td>
<td></td>
</tr>
<tr>
<td>Alpass, 87</td>
<td></td>
</tr>
<tr>
<td>Alpass filter, 87</td>
<td></td>
</tr>
<tr>
<td>Amosc, 82</td>
<td></td>
</tr>
<tr>
<td>Analog synthesizer, 204</td>
<td></td>
</tr>
<tr>
<td>And, 242</td>
<td></td>
</tr>
<tr>
<td>Annotation, audio, 194</td>
<td></td>
</tr>
<tr>
<td>Append, 236</td>
<td></td>
</tr>
<tr>
<td>Apply, 232</td>
<td></td>
</tr>
<tr>
<td>Apply-banded-bass-boost, 199</td>
<td></td>
</tr>
<tr>
<td>Apply-banded-delay, 199</td>
<td></td>
</tr>
<tr>
<td>Apply-banded-treble-boost, 199</td>
<td></td>
</tr>
<tr>
<td>Approximation, 84</td>
<td></td>
</tr>
<tr>
<td>Arc-sine-dist, 178</td>
<td></td>
</tr>
<tr>
<td>Arcsine distribution, 178</td>
<td></td>
</tr>
<tr>
<td>Aref, 235</td>
<td></td>
</tr>
<tr>
<td>Areson, 90</td>
<td></td>
</tr>
<tr>
<td>Args, 163</td>
<td></td>
</tr>
<tr>
<td>Arguments to a lisp function, 163</td>
<td></td>
</tr>
<tr>
<td>Arithmetic functions, 247</td>
<td></td>
</tr>
<tr>
<td>Arpeggiator, 20</td>
<td></td>
</tr>
<tr>
<td>Array from sound, 70</td>
<td></td>
</tr>
<tr>
<td>Array functions, 235</td>
<td></td>
</tr>
<tr>
<td>Array notation, sal, 55</td>
<td></td>
</tr>
<tr>
<td>Arrayp, 240</td>
<td></td>
</tr>
<tr>
<td>Articulation, 143, 146</td>
<td></td>
</tr>
<tr>
<td>Assoc, 237</td>
<td></td>
</tr>
<tr>
<td>Asterisk, 144</td>
<td></td>
</tr>
<tr>
<td>At, 101</td>
<td></td>
</tr>
<tr>
<td>At transformation, 31</td>
<td></td>
</tr>
<tr>
<td>At, sal, 54</td>
<td></td>
</tr>
<tr>
<td>At-abs, 101</td>
<td></td>
</tr>
<tr>
<td>At-abs, sal, 54</td>
<td></td>
</tr>
<tr>
<td>Atan, 248</td>
<td></td>
</tr>
<tr>
<td>Atom, 239</td>
<td></td>
</tr>
<tr>
<td>Atone, 89</td>
<td></td>
</tr>
<tr>
<td>Attributes, 143</td>
<td></td>
</tr>
<tr>
<td>Audacity, 194</td>
<td></td>
</tr>
<tr>
<td>Audio annotation, 194</td>
<td></td>
</tr>
<tr>
<td>Audio markers, 106</td>
<td></td>
</tr>
<tr>
<td>Audio output device, 132</td>
<td></td>
</tr>
<tr>
<td>Autocorrelation of vector data, 195</td>
<td></td>
</tr>
<tr>
<td>Automatic gain control, 196</td>
<td></td>
</tr>
<tr>
<td>Autonorm-off, 106</td>
<td></td>
</tr>
<tr>
<td>Autonorm-on, 106</td>
<td></td>
</tr>
<tr>
<td>Average, 115</td>
<td></td>
</tr>
<tr>
<td>Backquote, 232</td>
<td></td>
</tr>
<tr>
<td>Backward, 198</td>
<td></td>
</tr>
<tr>
<td>Baktrace, 246</td>
<td></td>
</tr>
<tr>
<td>Banded bass boost, 199</td>
<td></td>
</tr>
<tr>
<td>Banded delay, 199</td>
<td></td>
</tr>
<tr>
<td>Banded treble boost, 199</td>
<td></td>
</tr>
<tr>
<td>Topic</td>
<td>Page</td>
</tr>
<tr>
<td>--------------------------------------------</td>
<td>------</td>
</tr>
<tr>
<td>Bandfx.lsp</td>
<td>199</td>
</tr>
<tr>
<td>Bandpass filter</td>
<td>89</td>
</tr>
<tr>
<td>Bandpass2</td>
<td>91</td>
</tr>
<tr>
<td>Bartok</td>
<td>150</td>
</tr>
<tr>
<td>Begin</td>
<td>55</td>
</tr>
<tr>
<td>Behavioral abstraction</td>
<td>27</td>
</tr>
<tr>
<td>Behaviors</td>
<td>76</td>
</tr>
<tr>
<td>Bell sound</td>
<td>20</td>
</tr>
<tr>
<td>Bernoulli distribution</td>
<td>180</td>
</tr>
<tr>
<td>Bernoulli-dist</td>
<td>180</td>
</tr>
<tr>
<td>Beta distribution</td>
<td>179</td>
</tr>
<tr>
<td>Beta-dist</td>
<td>179</td>
</tr>
<tr>
<td>Big endian</td>
<td>260</td>
</tr>
<tr>
<td>Bigendianp</td>
<td>260</td>
</tr>
<tr>
<td>Bilateral exponential distribution</td>
<td>175</td>
</tr>
<tr>
<td>Bilateral-exponential-dist</td>
<td>175</td>
</tr>
<tr>
<td>Binary files</td>
<td>256</td>
</tr>
<tr>
<td>Binomial distribution</td>
<td>180</td>
</tr>
<tr>
<td>Binomial-dist</td>
<td>180</td>
</tr>
<tr>
<td>Biquad</td>
<td>90</td>
</tr>
<tr>
<td>Biquad-m</td>
<td>90</td>
</tr>
<tr>
<td>Bitwise logical functions</td>
<td>249</td>
</tr>
<tr>
<td>Blank</td>
<td>144</td>
</tr>
<tr>
<td>Block</td>
<td>245</td>
</tr>
<tr>
<td>Both-case-p</td>
<td>252</td>
</tr>
<tr>
<td>Boundp</td>
<td>241</td>
</tr>
<tr>
<td>Bowed</td>
<td>95</td>
</tr>
<tr>
<td>Bowed-freq</td>
<td>95</td>
</tr>
<tr>
<td>Brass sound</td>
<td>20</td>
</tr>
<tr>
<td>Break</td>
<td>225</td>
</tr>
<tr>
<td>Break button</td>
<td>22</td>
</tr>
<tr>
<td>Browse button</td>
<td>22</td>
</tr>
<tr>
<td>Browser</td>
<td>23</td>
</tr>
<tr>
<td>Browser, jnyqide</td>
<td>23</td>
</tr>
<tr>
<td>Build-harmonic</td>
<td>14, 78</td>
</tr>
<tr>
<td>Button</td>
<td>136</td>
</tr>
<tr>
<td>Button bar</td>
<td>22</td>
</tr>
<tr>
<td>Buzz</td>
<td>83</td>
</tr>
<tr>
<td>Call command</td>
<td>157</td>
</tr>
<tr>
<td>Car</td>
<td>235</td>
</tr>
<tr>
<td>Case</td>
<td>144, 243</td>
</tr>
<tr>
<td>Case-insensitive</td>
<td>52</td>
</tr>
<tr>
<td>Catch</td>
<td>243</td>
</tr>
<tr>
<td>Cauchy distribution</td>
<td>177</td>
</tr>
<tr>
<td>Cauchy-dist</td>
<td>177</td>
</tr>
<tr>
<td>Cdr</td>
<td>236</td>
</tr>
<tr>
<td>Cerror</td>
<td>246</td>
</tr>
<tr>
<td>Change directory</td>
<td>256</td>
</tr>
<tr>
<td>Char</td>
<td>252</td>
</tr>
<tr>
<td>Char-code</td>
<td>253</td>
</tr>
<tr>
<td>Char-downcase</td>
<td>253</td>
</tr>
<tr>
<td>Char-equal</td>
<td>254</td>
</tr>
<tr>
<td>Char-greaterp</td>
<td>254</td>
</tr>
<tr>
<td>Char-int</td>
<td>253</td>
</tr>
<tr>
<td>Char-lessp</td>
<td>254</td>
</tr>
<tr>
<td>Char-not-equal</td>
<td>254</td>
</tr>
<tr>
<td>Char-not-greaterp</td>
<td>254</td>
</tr>
<tr>
<td>Char-not-lessp</td>
<td>254</td>
</tr>
<tr>
<td>Char-upcase</td>
<td>253</td>
</tr>
<tr>
<td>Char/=</td>
<td>253</td>
</tr>
<tr>
<td>Char&lt;</td>
<td>253</td>
</tr>
<tr>
<td>Char&lt;=</td>
<td>253</td>
</tr>
<tr>
<td>Char=&gt;</td>
<td>253</td>
</tr>
<tr>
<td>Char&gt;=</td>
<td>253</td>
</tr>
<tr>
<td>Character functions</td>
<td>252</td>
</tr>
<tr>
<td>Characterp</td>
<td>240</td>
</tr>
<tr>
<td>Chdir, sal</td>
<td>56</td>
</tr>
<tr>
<td>Chorus</td>
<td>93, 116, 123, 198</td>
</tr>
<tr>
<td>Chowning, john</td>
<td>20</td>
</tr>
<tr>
<td>Clarinet</td>
<td>93, 94</td>
</tr>
<tr>
<td>Clarinet sound</td>
<td>20</td>
</tr>
<tr>
<td>Clarinet-all</td>
<td>94</td>
</tr>
<tr>
<td>Clarinet-freq</td>
<td>93</td>
</tr>
<tr>
<td>Class</td>
<td>230</td>
</tr>
<tr>
<td>Class class</td>
<td>230</td>
</tr>
<tr>
<td>Clean-up</td>
<td>246</td>
</tr>
<tr>
<td>Clip</td>
<td>43, 97, 115</td>
</tr>
<tr>
<td>Clipping</td>
<td>131</td>
</tr>
<tr>
<td>Clipping repair</td>
<td>197</td>
</tr>
<tr>
<td>Clock</td>
<td>155</td>
</tr>
<tr>
<td>Clock command</td>
<td>155</td>
</tr>
<tr>
<td>Close</td>
<td>256</td>
</tr>
<tr>
<td>Close-slider-panel</td>
<td>135</td>
</tr>
<tr>
<td>Co-termination</td>
<td>114</td>
</tr>
<tr>
<td>Code-char</td>
<td>253</td>
</tr>
<tr>
<td>Comb</td>
<td>88</td>
</tr>
<tr>
<td>Comb filter</td>
<td>88</td>
</tr>
<tr>
<td>Combination</td>
<td>103</td>
</tr>
<tr>
<td>Command line</td>
<td>12</td>
</tr>
<tr>
<td>Command loop</td>
<td>224</td>
</tr>
<tr>
<td>Commas</td>
<td>152</td>
</tr>
<tr>
<td>Comment</td>
<td>144</td>
</tr>
<tr>
<td>Comments</td>
<td>52</td>
</tr>
<tr>
<td>Compose</td>
<td>115</td>
</tr>
<tr>
<td>Compress</td>
<td>196</td>
</tr>
<tr>
<td>Compress-map</td>
<td>196</td>
</tr>
</tbody>
</table>
Compressor, 72
Concatenate strings, 251
Cond, 242
Conditional expression, sal, 55
Configure nyquist, 10
Congen, 88
Cons, 236
Console, xlisp, 261
Consp, 240
Const, 77
Constant function, 77
Continue, 246
Continuous-control-warp, 101
Continuous-sound-warp, 101
Contour generator, 88
Control, 76
Control change, 153
Control characters, xlisp, 224
Control constructs, 242
Control panel, 135
Control-a, 106
Control-srate-abs, 102
Control-warp, 78
Convert flonum to fixnum, 75
Convert sound to array, 70
Convolution, 88
Convolution tutorial, 89
Convolve, 88
Copier pattern, 171
Correlation, 194
Cos, 248
Cosine, vector, 195
Cue, 76
Cue-file, 76
Current-path, 164
Curve fitting, 195
Cxxr, 236
Cxxrx, 236
Cxxxx, 236
Cycle pattern, 168
Darwinremotesoc, 138
Data types, 225
Db-average, 196
Db-to-linear, 72
Db-to-vel, 72
Db0, 18
Db1, 18
Db10, 18
Debugging, 71, 112, 162, 246
Decf, 164
Decrement, 164
Default, 149
Default durations, 148
Default sample rate, 34
Default sound file directory, 106
Default time, 144
Define function, 56
Define variable, 56
Defining behaviors, 32
Defmacro, 234
Defun, 234
Delay, 89
Delay, variable, 116
Delete, 239
Delete-if, 239
Delete-if-not, 239
Demos, bell sound, 20
Demos, convolution, 89
Demos, distortion, 90
Demos, drum machine, 20
Demos, drum sound, 20
Demos, fft, 140
Demos, fm, 46
Demos, fm synthesis, 20
Demos, formants, 20
Demos, gong sound, 20
Demos, lpc, 159
Demos, midi, 142
Demos, multiple band effects, 199
Demos, piano, 195
Demos, pitch change, 115
Demos, probability distributions, 174
Demos, rhythmic pattern, 20
Demos, ring modulation, 17
Demos, sample-by-sample, 20
Demos, scratch tutorial, 46
Demos, shepard tones, 90
Demos, slider control, 135
Demos, spectral analysis of a chord, 20
Demos, voice synthesis, 90
Demos, wind sound, 48
Derivative, 80
Describe, 192
Destructive list functions, 238
Developing code, 162
Devices, 132
Index

Diff, 105
Difference, 193
Difference of sounds, 105
Difference of vector data, 195
Digit-char, 253
Digit-char-p, 253
Directory, default sound file, 106
Directory listing, 256
Display statement, sal, 61
Distortion tutorial, 90
Distributions, probability, 174
Division, 99
Do, 244
Do*, 244
Dolby pro-logic, 202
Dolby surround, 201
Dolist, 244
Doppler effect, 203
Dot, 145
Dotimes, 244
Dotted durations, 18
Dribble, 259
Drum, 203
Drum machine, 20, 203
Drum samples, 20
Drum sound, 20
Drum-loop, 204
Dsp in lisp, 20, 48
Dtmf, 201
Dtmf-tone, 201
Dubugging, 116
Duration, 143, 145
Duration notation, 18
Duration of another sound, 114
Dx7, 146
Dynamic markings, 147

Echo, 89
Echoenabled, 261
Editor for envelopes, 23
Effect, chorus, 93, 123, 198
Effect, flange, 198
Effect, pitch shift, 93, 123
Effect, reverberation, 92, 123, 201
Effect, stereo, 201
Effect, stereo pan, 202
Effect, swap channels, 202
Effect, widen, 201
Effects, phaser, 198

Eighth note, 18, 145
Elapsed audio time, 106
Emacs, using nyquist with, 260
Empty list, 53
End, 55
End command, 157
Endian, 260
Endless tones, 20
Endp, 240
Env, 16, 77
Env-note, 16
Envedit button, 22
Envelope, 16
Envelope editor, 23
Envelope follower, 72, 116
Envelope generator, 88
Envelopes, 16
Environment, 27
Environment variables, 259
Eq, 241
Eq button, 22
Eq-band, 91
Eq-highshelf, 91
Eq-lowshelf, 91
Eql, 242
Equal, 242
Equalization, 91, 197
Equalization editor, 24
Error, 246
Error handling, 246
Errors, 8
Errset, 246
Estimate frequency, 100
Eval, 232
Eval pattern, 172
Evalhook, 246
Evaluation functions, 232
Evaluator, 226
Evenp, 241
Event-dur, 185
Event-end, 186
Event-expression, 186
Event-get-attr, 186
Event-has-attr, 186
Event-set-attr, 186
Event-set-dur, 186
Event-set-expression, 186
Event-set-time, 185

v 3.12 267
<table>
<thead>
<tr>
<th>Term</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Event-time</td>
<td>185</td>
</tr>
<tr>
<td>Exclamation point</td>
<td>144</td>
</tr>
<tr>
<td>Exec statement, sal</td>
<td>57</td>
</tr>
<tr>
<td>Exit</td>
<td>260</td>
</tr>
<tr>
<td>Exit statement, sal</td>
<td>63</td>
</tr>
<tr>
<td>Exp</td>
<td>249</td>
</tr>
<tr>
<td>Exp-dec</td>
<td>77</td>
</tr>
<tr>
<td>Expand</td>
<td>259</td>
</tr>
<tr>
<td>Exponent</td>
<td>165</td>
</tr>
<tr>
<td>Exponential</td>
<td>98</td>
</tr>
<tr>
<td>Exponential distribution</td>
<td>175</td>
</tr>
<tr>
<td>Exponential envelope</td>
<td>77</td>
</tr>
<tr>
<td>Exponential-dist</td>
<td>175</td>
</tr>
<tr>
<td>Expr-get-attr</td>
<td>186</td>
</tr>
<tr>
<td>Expr-has-attr</td>
<td>186</td>
</tr>
<tr>
<td>Expr-set-attr</td>
<td>186</td>
</tr>
<tr>
<td>Expression pattern</td>
<td>172</td>
</tr>
<tr>
<td>Expressions, sal</td>
<td>52</td>
</tr>
<tr>
<td>Expt</td>
<td>249</td>
</tr>
<tr>
<td>Extending xlisp</td>
<td>219</td>
</tr>
<tr>
<td>Extract</td>
<td>102</td>
</tr>
<tr>
<td>Extract-abs</td>
<td>102</td>
</tr>
<tr>
<td>F (adagio dynamic)</td>
<td>147</td>
</tr>
<tr>
<td>F (adagio flat)</td>
<td>145</td>
</tr>
<tr>
<td>Fast fourier transform tutorial</td>
<td>140</td>
</tr>
<tr>
<td>Fboundp</td>
<td>241</td>
</tr>
<tr>
<td>Feedback fm oscillator</td>
<td>82</td>
</tr>
<tr>
<td>Feedback-delay</td>
<td>89</td>
</tr>
<tr>
<td>Feel factor</td>
<td>190</td>
</tr>
<tr>
<td>Ff (adagio dynamic)</td>
<td>147</td>
</tr>
<tr>
<td>Fff (adagio dynamic)</td>
<td>147</td>
</tr>
<tr>
<td>Fft</td>
<td>140</td>
</tr>
<tr>
<td>Fft tutorial</td>
<td>140</td>
</tr>
<tr>
<td>File access limit</td>
<td>12</td>
</tr>
<tr>
<td>File i/o functions</td>
<td>255, 261</td>
</tr>
<tr>
<td>Filep</td>
<td>241</td>
</tr>
<tr>
<td>Filter example</td>
<td>48</td>
</tr>
<tr>
<td>Finally clause, sal</td>
<td>60</td>
</tr>
<tr>
<td>Find-in-xlisp-path</td>
<td>256</td>
</tr>
<tr>
<td>Find string</td>
<td>250</td>
</tr>
<tr>
<td>Fir filter</td>
<td>88</td>
</tr>
<tr>
<td>First</td>
<td>236</td>
</tr>
<tr>
<td>First derivative</td>
<td>80</td>
</tr>
<tr>
<td>Flange</td>
<td>198</td>
</tr>
<tr>
<td>Flange effect</td>
<td>198</td>
</tr>
<tr>
<td>Flat</td>
<td>145</td>
</tr>
<tr>
<td>Flate</td>
<td>255</td>
</tr>
<tr>
<td>Flatsize</td>
<td>255</td>
</tr>
<tr>
<td>Flet</td>
<td>243</td>
</tr>
<tr>
<td>Float</td>
<td>247</td>
</tr>
<tr>
<td>Floatp</td>
<td>240</td>
</tr>
<tr>
<td>Flute</td>
<td>95</td>
</tr>
<tr>
<td>Flute sound</td>
<td>20</td>
</tr>
<tr>
<td>Flute-all</td>
<td>95</td>
</tr>
<tr>
<td>Flute-freq</td>
<td>95</td>
</tr>
<tr>
<td>Fm synthesis</td>
<td>46</td>
</tr>
<tr>
<td>Fm voices</td>
<td>20</td>
</tr>
<tr>
<td>Fmfb</td>
<td>82</td>
</tr>
<tr>
<td>Fmfo</td>
<td>77</td>
</tr>
<tr>
<td>Fmosc</td>
<td>82</td>
</tr>
<tr>
<td>Fn button</td>
<td>22</td>
</tr>
<tr>
<td>Follow</td>
<td>72</td>
</tr>
<tr>
<td>Follower</td>
<td>116</td>
</tr>
<tr>
<td>Force-srate</td>
<td>77</td>
</tr>
<tr>
<td>Format</td>
<td>255</td>
</tr>
<tr>
<td>Fourth</td>
<td>236</td>
</tr>
<tr>
<td>Frequency analysis</td>
<td>100</td>
</tr>
<tr>
<td>Frequency modulation</td>
<td>45</td>
</tr>
<tr>
<td>Full path name</td>
<td>164</td>
</tr>
<tr>
<td>Funcall</td>
<td>232</td>
</tr>
<tr>
<td>Function</td>
<td>232</td>
</tr>
<tr>
<td>Function calls, sal</td>
<td>55</td>
</tr>
<tr>
<td>Function, sal</td>
<td>56</td>
</tr>
<tr>
<td>Fundamental frequency estimation</td>
<td>100</td>
</tr>
<tr>
<td>Gain</td>
<td>196</td>
</tr>
<tr>
<td>Gamma-dist</td>
<td>175</td>
</tr>
<tr>
<td>Gate</td>
<td>73, 117</td>
</tr>
<tr>
<td>Gaussian distribution</td>
<td>179</td>
</tr>
<tr>
<td>Gaussian-dist</td>
<td>179</td>
</tr>
<tr>
<td>Gc</td>
<td>259</td>
</tr>
<tr>
<td>Gcd</td>
<td>248</td>
</tr>
<tr>
<td>Gen05</td>
<td>87</td>
</tr>
<tr>
<td>Gensym</td>
<td>234</td>
</tr>
<tr>
<td>Geometric distribution</td>
<td>180</td>
</tr>
<tr>
<td>Geometric-dist</td>
<td>180</td>
</tr>
<tr>
<td>Get</td>
<td>235</td>
</tr>
<tr>
<td>Get-duration</td>
<td>75</td>
</tr>
<tr>
<td>Get-env</td>
<td>259</td>
</tr>
<tr>
<td>Get-lambda-expression</td>
<td>233</td>
</tr>
<tr>
<td>Get-loud</td>
<td>75</td>
</tr>
<tr>
<td>Get-output-stream-list</td>
<td>258</td>
</tr>
<tr>
<td>Get-output-stream-string</td>
<td>258</td>
</tr>
<tr>
<td>Get-slider-value</td>
<td>136</td>
</tr>
<tr>
<td>Get-sustain</td>
<td>75</td>
</tr>
<tr>
<td>Get-temp-path</td>
<td>256</td>
</tr>
<tr>
<td>Get-transpose</td>
<td>75</td>
</tr>
</tbody>
</table>
Get-user, 256
Get-warp, 75
Get char, 256
Getenv, 259
Global variables, 130
Global variables, sal, 56
Gnuplot, 194
Go, 245
Gong sounds, 20
Granular synthesis, 200
Graphical envelope editor, 23
Graphical equalization, 24
Graphical equalizer, 197
Grind, 163

H, 18
H (adagio half note), 145
Half note, 18, 145
Harmonic, 78
Hash, 234
Hd, 18
Header file format, 219
Heap pattern, 170
High-pass filter, 89
Highpass2, 91
Highpass4, 92
Highpass6, 92
Highpass8, 92
Histogram, 194
Hp, 89
Ht, 19
Hyperbolic-cosine-dist, 177
Hz-to-step, 73
Hzosc, 81

I, 18
I (adagio eight note), 145
Iannis xenakis, 24
Id, 18
Ide, 11
If, 242
If statement, sal, 57
Ifft, 140
Incf, 163
Increment, 163
Info, 259
Info button, 22
Input from a file, 261
Input/output functions, 254
Limit, run-time, 12
Limiter, 72
Line pattern, 169
Linear algebra, 195
Linear distribution, 175
Linear interpolation, 192
Linear prediction, 159
Linear prediction tutorial, 159
Linear regression, 195
Linear-dist, 175
Linear-to-db, 73
Linear-to-vel, 74
Lisp button, 22
Lisp dsp, 20, 48
Lisp include files, 222
List, 236
List functions, 235
List output devices, 132
List directory, 256
Listdir, 256
Listing of lisp function, 163
Lisp, 240
Little endian, 260
Lmf, 18
Lmf (adagio dynamic), 147
Lmp, 18
Lmp (adagio dynamic), 147
Load, 259
Load button, 22
Load file conditionally, 164
Load statement, sal, 58
Local-to-global, 75
Log, 74
Log function, 74
Logand, 249
Logical-stop, 67
Logior, 249
Logistic distribution, 178
Logistic-dist, 178
Lognot, 249
Log algorithm, 98
Logxor, 249
Loop, 244
Loop examples, sal, 60
Loop statement, sal, 59
Looping constructs, 244
Loud, 102
Loud-abs, 102
Loudness, 143, 147
Low-frequency oscillator, 77
Low-pass filter, 89, 123
Lower-case-p, 252
Lowpass2, 90
Lowpass4, 92
Lowpass6, 92
Lowpass8, 92
Lp, 18, 89
Lp (adagio dynamic), 147
Lpc, 159
Lpc tutorial, 159
Lpc-frame-err, 159, 160
Lpc-frame-filter-coefs, 159, 160
Lpc-frame-rms1, 159, 160
Lpc-frame-rms2, 159, 160
Lpp, 18
Lpp (adagio dynamic), 147
Lppp, 18
Lppp (adagio dynamic), 147
Lpreson, 160
Lpslider, 136
M (adagio control), 153
Macroexpand, 233
Macroexpand-1, 233
Macrolet, 243
Magnitude spectrum plot, 111
Make-accumulate, 171
Make-accumulation, 171
Make-array, 235
Make-button, 136
Make-copier, 171
Make-cycle, 168
Make-eval, 172
Make-heap, 170
Make-length, 173
Make-line, 169
Make-lpanal-iterator, 159
Make-lpc-file-iterator, 160
Make-markov, 173
Make-palindrome, 170
Make-product, 172
Make-random, 169
Make-slider, 135
Make-slider-panel, 135
Make-string-input-stream, 258
Make-string-output-stream, 258
Make-sum, 172
Make-symbol, 234
Make-window, 173
Makeetable, 77
Mandolin, 96
Manipulation of scores, 185
Mapc, 237
Mapcar, 238
Mapl, 238
Maplist, 238
Mark button, 22
Markers, audio, 106
Markov analysis, 174
Markov pattern, 173
Markov-create-rules, 174
Max, 248
Max, statistics, 194
Maximum, 98, 248
Maximum amplitude, 43, 118
Maximum of two sounds, 118
Mean, 194
Mean of vector, 195
Median of vector data, 195
Median, statistics, 194
Member, 236
Memory limit, 12, 72
Memory usage, 12, 71, 72
Mf (adagio dynamic), 147
Middle c, 145
Midi, 142
Midi clock, 155
Midi file, 190, 191
Midi file readr, 142
Midi file synthesis, 143
Midi file writer, 143
Midi program, 147
Midi sequence, 142
Midi tutorial, 142
Midi-show, 200
Midi-show-file, 200
Mikrokosmos, 150
Min, 248
Min, statistics, 194
Minimoog, 204
Minimum, 98, 248
Minusp, 241
Mix, 105
Mix to file, 110
Mkwave, 14
Modalbar, 96
Modulation wheel, 153
Modulo (rem, 247
Mono to stereo, 201
Moog, 204
Moving average, 115
Mp (adagio dynamic), 147
Mult, 16, 78, 105
Multichannel sounds, 67
Multiple band effects, 199
Multiple commands, 152
Multiple tempi, 154
Multiplication, 118
Multiply signals, 105
My-note, 15
N (adagio next), 146
Natural, 145
Natural log, 98
Nband, 197
Nband-range, 197
Nconc, 239
Nested transformations, 32
Newfile button, 22
Next adagio command, 146
Next in pattern, 167
Next pattern, 167
Nintendo wiimote, 138
Noise, 99
Noise gate, 117
Noise-gate, 73
Norm of vector, 195
Normalization, 43
Not, 240
Not enough memory for normalization, 43
Notch filter, 90
Notch2, 91
Note list, 105
Nrev, 92
Nstring-downcase, 251
Nstring-upcase, 250
Nth, 237
Nthcdr, 237
Null, 240
Numberp, 240
Ny:all, 19
Nyquistide, 11
O (adagio control), 153
Object, 230
Object class, 230
Objectp, 241
Objects, 229
Octave specification, 145
Oddp, 241
Offset, 190
Offset to a sound, 115
Omissions, 8
Oneshot, 118
Open, 256
Open sound control, 138
Open-binary, 256
Openfile button, 22
Or, 242
Osc, 14, 81, 138
Osc-enable, 138
Osc-note, 98
Osc-pulse, 82
Osc-saw, 82
Osc-tri, 82
Output device, 132
Output samples to file, 107
Output to a file, 261
Overlap, 103
Overwrite samples, 110
P (adagio dynamic), 147
P (adagio pitch), 145
Palindrome pattern, 170
Pan, 78, 201
Pan, stereo, 202
Parameters, keyword, 183
Params-scale, 192
Params-transpose, 192
Partial, 81
Path, current, 164
Pattern objects, 167
Pattern, accumulate, 171
Pattern, accumulation, 171
Pattern, copier, 171
Pattern, cycle, 168
Pattern, eval, 172
Pattern, expression, 172
Pattern, heap, 170
Pattern, length, 173
Pattern, line, 169
Pattern, markov, 173
Pattern, palindrome, 170
Pattern, product, 172
Pattern, random, 169
Pattern, sum, 172
Pattern, window, 173
Pattern-class, 167
Patternp, 192
Patterns, 166
Peak, 117
Peak amplitude, 43
Peak, maximum amplitude, 117
Pearson correlation, 194
Peek, 260
Peek-char, 256
Period estimation, 100
Phase vocoder, 96
Phaser, 198
Phasevocoder, 96, 126
Physical model, 20
Physical models, 93
Piano synthesizer, 195
Piano synthesizer tutorial, 195
Piano-midi, 195
Piano-midi2file, 195
Piano-note, 195
Piano-note-2, 195
Piece-wise, 84
Piece-wise linear, 118
Pitch, 143, 145
Pitch bend, 153
Pitch detection, 100
Pitch notation, 19
Pitch shift, 93, 97, 123
Pitch shifting, 115
Pitchshift, 93
Pl-center, 202
Pl-doppler, 203
Pl-left, 202
Pl-pan2d, 202
Pl-position, 202
Pl-rear, 202
Pl-right, 202
Play, 14, 105
Play in reverse, 198
Play statement, sal, 61
Play-file, 106
Plot, 111
Plot statement, sal, 61
Plots, 194
<table>
<thead>
<tr>
<th>Topic</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pluck, 83</td>
<td>273</td>
</tr>
<tr>
<td>Plucked string, 83</td>
<td>273</td>
</tr>
<tr>
<td>Plusp, 241</td>
<td>273</td>
</tr>
<tr>
<td>Poisson distribution, 180</td>
<td>273</td>
</tr>
<tr>
<td>Poisson-dist, 180</td>
<td>273</td>
</tr>
<tr>
<td>Poke, 260</td>
<td>273</td>
</tr>
<tr>
<td>Polyrhythm, 154</td>
<td>273</td>
</tr>
<tr>
<td>Pop, 164</td>
<td>273</td>
</tr>
<tr>
<td>Portamento switch, 153</td>
<td>273</td>
</tr>
<tr>
<td>Portaudio, 132</td>
<td>273</td>
</tr>
<tr>
<td>Power, 165</td>
<td>273</td>
</tr>
<tr>
<td>Pp (adagio dynamic), 147</td>
<td>273</td>
</tr>
<tr>
<td>Ppp (adagio dynamic), 147</td>
<td>273</td>
</tr>
<tr>
<td>Pprev, 92</td>
<td>273</td>
</tr>
<tr>
<td>Predicate functions, 239</td>
<td>273</td>
</tr>
<tr>
<td>Preset, 147</td>
<td>273</td>
</tr>
<tr>
<td>Prin1, 254</td>
<td>273</td>
</tr>
<tr>
<td>Princ, 254</td>
<td>273</td>
</tr>
<tr>
<td>Print, 254</td>
<td>273</td>
</tr>
<tr>
<td>Print midi file, 200</td>
<td>273</td>
</tr>
<tr>
<td>Print statement, sal, 61</td>
<td>273</td>
</tr>
<tr>
<td>Probability distributions, 174</td>
<td>273</td>
</tr>
<tr>
<td>Prod, 78</td>
<td>273</td>
</tr>
<tr>
<td>Product, 105</td>
<td>273</td>
</tr>
<tr>
<td>Product pattern, 172</td>
<td>273</td>
</tr>
<tr>
<td>Profile, 247</td>
<td>273</td>
</tr>
<tr>
<td>Profiling, 231</td>
<td>273</td>
</tr>
<tr>
<td>Prog, 244</td>
<td>273</td>
</tr>
<tr>
<td>Prog*, 244</td>
<td>273</td>
</tr>
<tr>
<td>Prog1, 245</td>
<td>273</td>
</tr>
<tr>
<td>Prog2, 245</td>
<td>273</td>
</tr>
<tr>
<td>Progn, 245</td>
<td>273</td>
</tr>
<tr>
<td>Program, 153</td>
<td>273</td>
</tr>
<tr>
<td>Program change, 146</td>
<td>273</td>
</tr>
<tr>
<td>Progv, 245</td>
<td>273</td>
</tr>
<tr>
<td>Prologic, 202</td>
<td>273</td>
</tr>
<tr>
<td>Property list functions, 235</td>
<td>273</td>
</tr>
<tr>
<td>Protected files, 12</td>
<td>273</td>
</tr>
<tr>
<td>Psetq, 233</td>
<td>273</td>
</tr>
<tr>
<td>Pulse oscillator, 82</td>
<td>273</td>
</tr>
<tr>
<td>Pulse-width modulation, 82</td>
<td>273</td>
</tr>
<tr>
<td>Push, 164</td>
<td>273</td>
</tr>
<tr>
<td>Putprop, 235</td>
<td>273</td>
</tr>
<tr>
<td>Pv-time-pitch, 97</td>
<td>273</td>
</tr>
<tr>
<td>Pwe, 86</td>
<td>273</td>
</tr>
<tr>
<td>Pwe-list, 86</td>
<td>273</td>
</tr>
<tr>
<td>Pwer, 87</td>
<td>273</td>
</tr>
</tbody>
</table>

v 3.12
Remove-if, 237
Remove-if-not, 237
Remprop, 235
Replace file samples, 110
Replay button, 22
Require-from, 164
Resample, 79
Resampling, 77, 115
Rescaling, 43
Resolution, 152
Reson, 89
Rest, 99, 236
Restore, 259
Rests, 146
Return, 245
Return statement, sal, 62
Return-from, 245
Reverb, 92, 123, 201
Reverse, 236
Reverse, sound, 198
Ring modulation, 17
Risset, 20
Rms, 99, 115
Room, 259
Round, 75
Rplaca, 238
Rplacd, 238
Rrandom, 248
Run-time limit, 12
Sal, 51
S (adagio sharp), 145
S (adagio sixteenth note), 145
S-abs, 97
S-add-to, 110
S-exp, 98
S-log, 98
S-max, 43, 98
S-min, 43, 98
S-overwrite, 110
S-plot, 111
S-print-tree, 112
S-read, 109
S-read-reverse, 198
S-rest, 99
S-reverse, 198
S-save, 107
S-sqrt, 98
Sal and lisp, 63
Sal button, 22
Sal expressions, 52
Sample interpolation, 119
Sample rate, forcing, 77
Sampler, 83
Samples, 67, 70
Samples, reading, 69
Sampling rate, 33, 71, 74
Save, 259
Save samples to file, 107
Save-lpc-file, 160
Save-workspace, 192
Savefile button, 22
Saving sound files, 42
Sawtooth oscillator, 82
Sawtooth table, 132
Sawtooth wave, 15
Sax, 94
Sax-all, 94
Sax-freq, 94
Scale, 79
Scale, tuning, 19, 74
Scale-db, 79
Scale-srate, 79
Scan directory, 256
Score, 105
Score manipulation, 185
Score, musical, 16
Score-adjacent-events, 189
Score-append, 188
Score-apply, 189
Score-filter, 188
Score-filter-length, 189
Score-filter-overlap, 189
Score-gen, 183, 184
Score-get-begin, 188
Score-get-end, 189
Score-indexof, 190
Score-last-indexof, 190
Score-merge, 188
Score-must-have-begin-end, 189
Score-play, 189
Score-print, 189
Score-randomize-start, 190
Score-read, 191
Score-read-smf, 190
Score-repeat, 189
Score-scale, 187
Score-select, 188
Score-set-begin, 188
Score-set-end, 188
Score-shift, 187
Score-sort, 187
Score-sorted, 187
Score-stretch, 187
Score-stretch-to-length, 189
Score-sustain, 187
Score-transpose, 187
Score-voice, 188
Score-write, 191
Score-write-smf, 191
Scratch sound, 46
Sd, 18
Second, 236
Sections, adagio, 151
Security, 12
Seed, 248
Select output device, 132
Self, 231
Semicolon, adagio, 152
Seq, 104
Seq-create, 142
Seq-insert-note, 143
Seq-midi, 143
Seq-midi-sal, 143
Seq-read, 142
Seq-read-smf, 142
Seq-write, 143
Seq-write-smf, 143
Seqrep, 104
Sequence, 142
Sequence_example.htm, 16
Sequences, 15, 142
Sequential behavior, 29
Set, 233
Set intersection, 193
Set statement, sal, 62
Set union, 193
Set-control-srate, 74
Set-difference, 193
Set-logical-stop, 105
Set-pitch-names, 74
Set-sound-srate, 74
Setdir, 256
Setf, 233
Seti command, 157
Setq, 233
Setup nyquist, 10
Setup-console, 260
Setv command, 157
Sf-granulate, 200
Sf-info, 111
Shape, 90
Sharp, 145
Shepard tones, 20, 90
Shift-time, 79
Show midi file, 200
Show output devices, 132
Show-lpc-data, 160
Signal composition, 115, 119
Signal multiplication, 118
Signal-start, 67
Signal-stop, 67
Sim, 14, 104
Simrep, 104
Simultaneous behavior, 29
Sin, 248
Sine, 81
Sine table, 132
Siosc, 83
Sitar, 96
Sixteenth note, 18, 145
Sixtyfourth note, 145
Slider, 135, 136
Slider tutorial, 135
Slider-panel, 135
Slope, 80
Smooth, 80
Snd-abs, 114
Snd-add, 115
Snd-allpoles, 161
Snd-alpass, 120
Snd-alpasscv, 120
Snd-alpassvv, 120
Snd-amosc, 125
Snd-areson, 120
Snd-aresoncv, 121
Snd-aresonvc, 121
Snd-aresonvv, 121
Snd-atomel, 121
Snd-atomn, 121
Snd-avg, 115
Snd-bandedwg, 127
Snd-biquad, 121
Snd-bowed, 127
Snd-bowed-freq, 127
Snd-buzz, 125
Snd-chase, 121
Snd-clarinet, 127
Snd-clarinet-all, 127
Snd-clarinet-freq, 127
Snd-clip, 115
Snd-compose, 115
Snd-congen, 122
Snd-const, 112
Snd-convolve, 122
Snd-copy, 116
Snd-coterm, 114
Snd-delay, 122
Snd-delaycv, 122
Snd-down, 116
Snd-exp, 116
Snd-extent, 69
Snd-fetch, 69
Snd-fetch-array, 69
Snd-fit, 140
Snd-flatten, 70
Snd-flute, 127
Snd-flute-all, 128
Snd-flute-freq, 128
Snd-fmfb, 125
Snd-fmfbv, 125
Snd-fmosc, 125
Snd-follow, 116
Snd-from-array, 68
Snd-fromarraystream, 68
Snd-fromobject, 69
Snd-gate, 117
Snd-iff, 140
Snd-inverse, 117
Snd-length, 70
Snd-log, 117
Snd-lpanel, 161
Snd-lpreson, 161
Snd-mandolin, 128
Snd-max, 118
Snd-maxamp, 70
Snd-maxv, 118
Snd-modalbar, 128
Snd-multiseq, 129
Snd-normalize, 118
Snd-offset, 115
Snd-oneshot, 118
Snd-osc, 125
Snd-overwrite, 113
Snd-partial, 125
Snd-phase vocoder, 126
Snd-play, 70
Snd-pluck, 125
Snd-print, 71
Snd-print-tree, 70, 112
Snd-prod, 118
Snd-pwl, 118
Snd-quantize, 119
Snd-read, 112
Snd-recip, 119
Snd-resample, 119
Snd-resamplenv, 119
Snd-reson, 122
Snd-resoncv, 123
Snd-resonvc, 123
Snd-resonvv, 123
Snd-sampler, 126
Snd-samples, 70
Snd-save, 113
Snd-sax, 128
Snd-sax-all, 128
Snd-sax-freq, 128
Snd-scale, 119
Snd-seq, 129
Snd-set-latency, 75
Snd-set-logical-stop, 71
Snd-set-max-audio-mem, 72
Snd-shape, 119
Snd-sine, 126
Snd-siosc, 126
Snd-sitar, 129
Snd-slider, 137
Snd-sqrt, 114
Snd-srate, 71
Snd-sref, 71
Snd-stkchorus, 123
Snd-stkpitchshift, 123
Snd-stkrev, 123
Snd-stop-time, 71
Snd-stoponzero, 138
Snd-t0, 71
Snd-tapf, 116
Snd-tapv, 116
Snd-time, 71
Snd-tone, 123
Snd-tonev, 124
Snd-trigger, 138
Snd-up, 119
Snd-white, 114
Snd-xform, 120
Snd-yin, 120
Snd-zero, 114
Soften-clipping, 197
Sort, 239
Sound, 76
sound
accessing point, 67
creating from array, 68
Sound browser, jnyqide, 23
Sound file directory default, 106
Sound file i/o, 105
Sound file info, 111
Sound from lisp data, 69
Sound-off, 107
Sound-on, 107
Sound-rate-abs, 102
Sound-warp, 80
Soundfilename, 111
Soundp, 71
Sounds, 66
Sounds vs. behaviors, 30
Span, 202
Spatialization, 201
Spec-plot, 111
Special command, 144
Spectral interpolation, 83
Spectrum plot, 111
Speed-dial, 201
Splines, 84
Sqrt, 249
Square oscillator, 82
Square root, 98, 114
Srate, 67
Sref, 67
Sref-inverse, 68
St, 19
Stacatto, 103
Staccato, 146
Stack trace, 246
Standard deviation, 194
Standard deviation of vector, 195
Standard midi file, 190
Statements, sal, 55
Statistics, 194
Stats, 71
Step-to-hz, 74
Stereo, 201
Stereo pan, 202
Stereo panning, 78
Stereo-chorus, 198
Stereooize, 201
Stk banded waveguide, 127
Stk bowed, 127
Stk bowed string, 95
Stk bowed-freq, 95
Stk chorus, 93, 123
Stk clarinet, 93, 94, 127
Stk flute, 95, 127, 128
Stk glass harmonica, 96
Stk jcreverb, 92
Stk mandolin, 128
Stk mandolon, 96
Stk modal bar, 96, 128
Stk nreverb, 92
Stk pitch shift, 93, 123
Stk prcreverb, 92
Stk reverb, 123
Stk sax, 94, 128
Stk sitar, 96, 129
Stk tibetan bowl, 96
Stk tuned bar, 96
Stk uniform bar, 96
Stkchorus, 93
Stochastic functions, 174
Stop-on-zero, 137
Strcat, 251
Stream, 250
Stream functions, 250
Stream stream functions, 258
String synthesis, 83
String-downcase, 250
String-equal, 252
String-greaterp, 252
String-left-trim, 250
String-lessp, 252
String-not-equal, 252
String-not-greaterp, 252
String-not-lessp, 252
String-right-trim, 250
String-search, 250
String-trim, 250
String-upcase, 250
String/=, 251
String<, 251
String<=, 251
String>=, 251
Stringp, 240
Sublis, 238
Subseq, 251
Subset, 193
Subsetp, 193
Subst, 238
Suggestions, 8
Sum, 105
Sum pattern, 172
Surround sound, 202
Sustain, 103
Sustain-abs, 103
Swap channels, 202
Swapchannels, 202
Symbol functions, 233
Symbol-function, 234
Symbol-name, 234
Symbol-plist, 234
Symbol-value, 234
Symbolp, 239
Symbols, 231
Synchronization, 155
System functions, 259

T, 144
T (adagio triplet), 145
T-test, statistics, 194
Table, 90
Table memory, 71
Tagbody, 245
Tan, 248
Tap, 116
Tapped delay, 92
Tapv, 92

Temp file, 256
Tempo, 144, 148
Temporary sound files directory, 106
Temporary files, 256
Terpri, 255
The format function, 255
The program feature, 244
Third, 236
Thirtysecond note, 145
Threshold, 118
Throw, 243
Time, 143, 144, 148
Time shift, sal, 54
Time stretch, 96
Time structure, 103
Time units, 152
Timed-seq, 105
Tone, 89
Top button, 22
Top-level, 246
Touch tone, 201
Trace, 246
Transformation environment, 27
Transformations, 27, 101
Transpose, 97, 103
Transpose-abs, 103
Triangle oscillator, 82
Triangle table, 133
Triangle wave, 15
Trigger, 138
Trill, 157
Triplet, 145
Triplet durations, 19
Truncate, 247
Tuba, 20
Tuning, 19, 74
Tutorial, fm, 46
Type-of, 260

U, 145
Undo, 21
Uniform random, 165
Uniform random, 248
Union, 193
Unless, 243
Untrace, 246
Unwind-protect, 243
Uotic, 24
Upper-case-p, 252
<table>
<thead>
<tr>
<th>Term</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>User name</td>
<td>256</td>
</tr>
<tr>
<td>V (adagio voice)</td>
<td>147</td>
</tr>
<tr>
<td>Variable delay</td>
<td>92, 116</td>
</tr>
<tr>
<td>Variable-resample function</td>
<td>115</td>
</tr>
<tr>
<td>Variance</td>
<td>194</td>
</tr>
<tr>
<td>Variance of vector data</td>
<td>195</td>
</tr>
<tr>
<td>Vector</td>
<td>235</td>
</tr>
<tr>
<td>Vector cosine</td>
<td>195</td>
</tr>
<tr>
<td>Vector difference</td>
<td>195</td>
</tr>
<tr>
<td>Vector median</td>
<td>195</td>
</tr>
<tr>
<td>Vector norm</td>
<td>195</td>
</tr>
<tr>
<td>Vectors</td>
<td>195</td>
</tr>
<tr>
<td>Vel-to-db</td>
<td>76</td>
</tr>
<tr>
<td>Vel-to-linear</td>
<td>76</td>
</tr>
<tr>
<td>Velocity</td>
<td>147</td>
</tr>
<tr>
<td>Vinal scratch</td>
<td>46</td>
</tr>
<tr>
<td>Vocal sound</td>
<td>20</td>
</tr>
<tr>
<td>Voice</td>
<td>143, 147</td>
</tr>
<tr>
<td>Voice synthesis</td>
<td>90</td>
</tr>
<tr>
<td>Volume</td>
<td>153</td>
</tr>
<tr>
<td>W</td>
<td>18</td>
</tr>
<tr>
<td>W (adagio whole note)</td>
<td>145</td>
</tr>
<tr>
<td>Warble</td>
<td>46</td>
</tr>
<tr>
<td>Warp</td>
<td>103</td>
</tr>
<tr>
<td>Warp-abs</td>
<td>103</td>
</tr>
<tr>
<td>Waveforms</td>
<td>15, 78</td>
</tr>
<tr>
<td>Waveshaping</td>
<td>90</td>
</tr>
<tr>
<td>Wavetables</td>
<td>15, 78</td>
</tr>
<tr>
<td>Wd</td>
<td>18</td>
</tr>
<tr>
<td>Welch’s t-test</td>
<td>194</td>
</tr>
<tr>
<td>Wg-glass-harm</td>
<td>96</td>
</tr>
<tr>
<td>Wg-tibetan-bowl</td>
<td>96</td>
</tr>
<tr>
<td>Wg-tuned-bar</td>
<td>96</td>
</tr>
<tr>
<td>Wg-uniform-bar</td>
<td>96</td>
</tr>
<tr>
<td>When</td>
<td>164, 242</td>
</tr>
<tr>
<td>While</td>
<td>164</td>
</tr>
<tr>
<td>Whole note</td>
<td>18, 145</td>
</tr>
<tr>
<td>Widen</td>
<td>201</td>
</tr>
<tr>
<td>Wii controller</td>
<td>138</td>
</tr>
<tr>
<td>Wind sound</td>
<td>48</td>
</tr>
<tr>
<td>Wind_tutorial.htm</td>
<td>48</td>
</tr>
<tr>
<td>Window pattern</td>
<td>173</td>
</tr>
<tr>
<td>Window initialization</td>
<td>260</td>
</tr>
<tr>
<td>With statement</td>
<td>63</td>
</tr>
<tr>
<td>Wood drum sound</td>
<td>20</td>
</tr>
<tr>
<td>Workspace</td>
<td>24, 191</td>
</tr>
<tr>
<td>Write adagio file</td>
<td>143</td>
</tr>
<tr>
<td>Write midi file</td>
<td>143</td>
</tr>
<tr>
<td>Write samples to file</td>
<td>107</td>
</tr>
<tr>
<td>Write-byte</td>
<td>258</td>
</tr>
<tr>
<td>Write-char</td>
<td>257</td>
</tr>
<tr>
<td>Write-float</td>
<td>257</td>
</tr>
<tr>
<td>Write-int</td>
<td>257</td>
</tr>
<tr>
<td>Wt</td>
<td>19</td>
</tr>
<tr>
<td>X (adagio control)</td>
<td>153</td>
</tr>
<tr>
<td>Xenakis</td>
<td>24</td>
</tr>
<tr>
<td>Xlisp command loop</td>
<td>224</td>
</tr>
<tr>
<td>Xlisp data types</td>
<td>225</td>
</tr>
<tr>
<td>Xlisp evaluator</td>
<td>226</td>
</tr>
<tr>
<td>Xlisp lexical conventions</td>
<td>226</td>
</tr>
<tr>
<td>Xmusic</td>
<td>166</td>
</tr>
<tr>
<td>Y (adagio control)</td>
<td>153</td>
</tr>
<tr>
<td>Yin</td>
<td>100</td>
</tr>
<tr>
<td>Z (adagio control)</td>
<td>153</td>
</tr>
<tr>
<td>Z (adagio program)</td>
<td>147</td>
</tr>
<tr>
<td>Zerop</td>
<td>241</td>
</tr>
</tbody>
</table>