Excerpt from

Designing Sound

Practical synthetic sound design for film, games
and interactive media using dataflow

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CHAPTER 1 Introduction to PDF Pure Data guide

This is a an excerpt from the textbook "Designing Sound". Several years ago when I discovered the amazing Pure Data software I instantly knew it would be the vehicle for a book I was planning about sound design. Synthesis and advanced processing opens up a world of possibilities, limited only by imagination. But how can these be taught? How can they even be expressed? Sound design is often labelled a 'Black Art', not because designers cling to secret knowledge, but because of its ineffability, the paucity of language and expression. Until I discovered Pure Data my repertoire of techniques for game, TV and radio sound effects was a mish mash of ideas spanning dozens of applications. To demonstrate them would need volumes of screenshots and extended writing about proprietary applications that would certainly change and render the instructions worthless. Here at last was a coherent framework that could express complex ideas in a way that anyone could read. It was like a story teller suddenly discovering the written word, or a mathematician who had never seen written notation before.

One massive strength of Pure Data is that it's open source software. That means it's maintained and updated by an army of individuals motivated only by their love of the software and its value to us all. In addition to my gratitude to Miller Puckette for the fact that Pure Data even exists I am absolutely indebted to the Pure Data community. This textbook would simply not exist without the enormous help I have received from that community. From the start it has been my intention to return that energy. I began in 2005 to write tutorials about making sound effects using Pure Data and publishing them on a website http://obiwannabe.co.uk/. The website is now approaching its one millionth unique visitor.

Eventually it became apparent that the goals of documenting Pure Data for use in sound design, and the goals of writing about sound design more generally would diverge. At that point I decided my repayment to the community would be in the form of a subset of the book that worked as a basic Pure Data manual. For those not able to afford a textbook, and for those not needing the entire treatment of sound design for interactive applications, I hope this abridged PDF will be a useful introduction for Pure Data users.

The remainder of this introduction remains as is, from the first edition of the published textbook. The book contains 650 pages of material including 30 practical exercises for creating sounds. If you like Pure Data or other dataflow environments as a tool and you're involved in sound for film, video games, radio, TV, theatre, or writing interactive software then I do suggest you check out "Designing Sound". I do not think you will be disappointed.

Wishing you much fun and lots of luck in all your projects.

Andy Farnell, 2008

This is a textbook for anyone who wishes to understand and create sound effects starting from nothing. It's about sound as a process rather than sound as data, a subject sometimes called "procedural audio". The thesis of this book is that any sound can be generated from first principles, guided by analysis and synthesis. An idea evolving from this is, in some ways, sounds so constructed are more realistic and useful than recordings because they capture behaviour. Although considerable work is required to create synthetic sounds with comparable realism to recordings the rewards are astonishing. Sounds which are impossible to record become accessible. Transformations are made available that cannot be achieved though any existing effects process. And fantastic sounds can be created by reasoned extrapolation. This considerably widens the palette of the traditional sound designer beyond mixing and effecting existing material to include constructing and manipulating virtual sound objects. By doing so the designer obtains something with a remarkable property, something that has deferred form. Procedural sound is a living sound effect that can run as computer code and be changed in real time according to unpredictable events. The advantage of this for video games is enormous, though it has equally exciting applications for animations and other modern media.

About the book

Aims

The aim is to explore basic principles of making ordinary, everyday sounds using a computer and easily available free software. We use the Pure Data (Pd) language to construct *sound objects*, which unlike recordings of sound can be used later in a flexible way. A practical, systematic approach to procedural audio is taught by example and supplemented with background knowledge to give a firm context. From here the technically inclined artist will be able to build their own sound objects for use in interactive applications and other projects. Although it is not intended to be a manual for Pure Data, a sufficient introduction to patching is provided to enable the reader to complete the exercises. References and external resources on sound and computer audio are provided. These include other important textbooks, websites, applications, scientific papers and research supporting the material developed here.

Audience

Modern sound designers working in games, film, animation and media where sound is part of an interactive process will all find this book useful. Designers using traditional methods but looking for a deeper understanding and finer degree of control in their work will likewise benefit. Music production, traditional recording, arrangement, mixdown or working from sample libraries is not covered. It is assumed you are already familiar these concepts and have the ability to use multi-track editors like Ardour and *Pro Tools*$^{\text{TM}}$, plus the other necessary parts of a larger picture of sound design. It isn't aimed at complete beginners, but great effort is made to ease though the steep learning curve of digital signal processing (DSP) and synthesis at a gentle pace. Students of digital audio, sound production, music technology, film and game sound, and developers of audio software should all find something of interest here. It will appeal to those who know a little programming, but previous programming skills are not a requirement.

Using the book

Requirements

This is not a complete introduction to Pure Data, nor a compendium of sound synthesis theory. A sound designer requires a wide background knowledge, experience, imagination and patience. A grasp of everyday physics is helpful in order to analyse and understand sonic processes. An ambitious goal of this text is to teach synthetic sound production using very little maths. Where possible I try to explain pieces of signal processing theory in only words and pictures. However, from time to time code or equations do appear to illustrate a point, particularly in the earlier theory chapters where formulas are given for reference.

Although crafting sound from numbers is an inherently mathematical process we are fortunate that tools exist which hide away messy details of signal programming to allow a more direct expression as visual code. To get the best from this book a serious student should embark upon supporting studies of digital audio and DSP theory. For a realistic baseline, familiarity with simple arithmetic, trigonometry, logic and graphs is expected.

Previous experience patching with Pure Data or Max/MSP will give you a head start, but even if you've never used it before the principles are easy to learn. Although Pure Data is the main vehicle for teaching this subject an attempt is made to discuss the principles in an application agnostic way. Some of the content is readable and informative without the need for other resources, but to make the best use of it you should work alongside a computer set up as an audio workstation and complete the practical examples. The minimum system requirements for most examples are a 500MHz computer with 256MB of RAM, a sound card, loudspeakers or headphones, and a copy of the Pure Data program. A simple wave file editor, such as Audacity, capable of handling Microsoft *.wav* or Mac *.au* formats will be useful.

Structure

Many of the examples follow a pattern. First we discuss the nature and physics of a sound and talk about our goals and constraints. Next we explore the theory and gather food for developing synthesis models. After choosing a set of methods, each example is implemented, proceeding through several stages of refinement to produce a Pure Data program for the desired sound. To make good use of space and avoid repeating material I will sometimes present only the details of a program which change. As an ongoing subtext we will discuss, analyse and refine the different synthesis techniques we use. So that you don't have to enter every Pure Data program by hand the examples are available on a CD ROM and online to download. There are audio examples to help you understand if Pure Data is not available.

Written Conventions

Pure Data is abbreviated as Pd, and since other similar DSP patcher tools exist you may like to take Pd as meaning "patch diagram" in the widest sense. For most commands, keyboard shortcuts are given as `CTRL+s`, `RETURN` and so forth. Note, for Mac users `CTRL` refers to the "command" key and where `right click` or `left click` are specified you should use the appropriate keyboard and click combination. Numbers are written as floating point decimals almost everywhere, especially where they refer to signals, as a constant reminder that all numbers are floats in Pd. In other contexts ordinary integers will be written as such. Graphs are provided to show signals. These are generally normalised -1.0 to $+1.0$, but absolute scales or values should not be taken too seriously unless the discussion focuses on them. Scales are often left out for the simplicity of showing just the signal. When we refer to a Pd object within text it will appear as a small container box, like $\boxed{\texttt{metro}}$. The contents of the box are the object name, in this case a metronome. The motto of Pd is "The diagram is the program". This ideal, upheld by its author Miller Puckette, makes Pd very interesting for publishing and teaching because you can implement the examples just by looking at the diagrams.

CHAPTER 2 Starting with Pure Data

SECTION 2.1

Pure Data

Pure Data is a visual signal programming language which makes it easy to construct programs to operate on signals. We are going to use it extensively in this textbook as a tool for sound design. The program is in active development and improving all the time. It is a free alternative to *Max/MSP*™ that many see as an improvement.

The primary application of Pure Data is processing sound, which is what it was designed for. However, it has grown into a general purpose signal processing environment with many other uses. Collections of video processing externals exist called *Gem*, *PDP* and *Gridflow* which can be used to create 3D scenes and manipulate 2D images. It has a great collection of interfacing objects, so you can easily attach joysticks, sensors and motors to prototype robotics or make interactive media installations. It is also a wonderful teaching tool for audio signal processing. Its economy of visual expression is a blessing: in other words it doesn't look too fancy, which makes looking at complex programs much easier on the eye. There is a very powerful idea behind "The diagram is the program". Each patch contains its complete state visually so you can reproduce any example just from the diagram. That makes it a visual description of sound.

The question is often asked "Is Pure Data a programming language?". The answer is yes, in fact it is a Turing complete language capable of doing anything that can be expressed algorithmically, but there are tasks such as building text applications or websites that Pure Data is ill suited to. It is a specialised programming language that does the job it was designed for very well, processing signals. It is like many other GUI frameworks or DSP environments which operate inside a "canned loop"[1] and are not truly open programming languages. There is a limited concept of iteration, programmatic branching, and conditional behaviour. At heart dataflow programming is very simple. If you understand object oriented programming, think of the objects as having methods which are called by data, and can only return data. Behind the scenes Pure Data is quite sophisticated. To make signal programming simple it hides away behaviour like

[1]A canned loop is used to refer to languages in which the real low level programmatic flow is handled by an interpreter that the user is unaware of

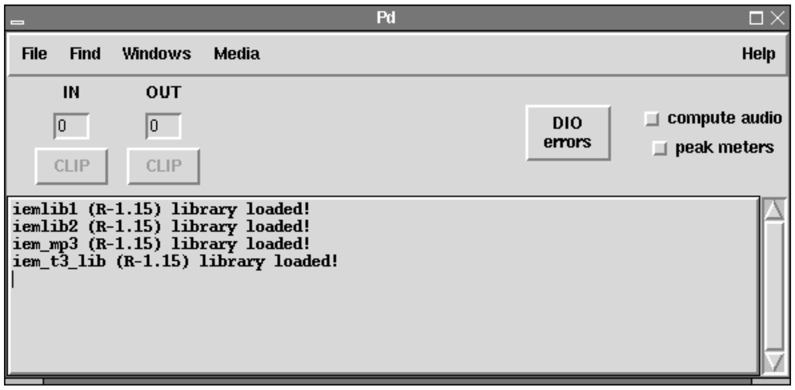
deallocation of deleted objects and manages the execution graph of a multi-rate DSP object interpreter and scheduler.

Installing and running Pure Data

Grab the latest version for your computer platform by searching the internet for it. There are versions available for Mac, Windows and Linux systems. On Debian based Linux systems you can easily install it by typing:

```
$ apt-get install puredata
```

Ubuntu and RedHat users will find the appropriate installer in their package management systems, and MacOSX or Windows users will find an installer program online. Try to use the most up to date version with libraries. The *pd-extended* build includes extra libraries so you don't need to install them separately. When you run it you should see a console window that looks something like Fig. 2.1.

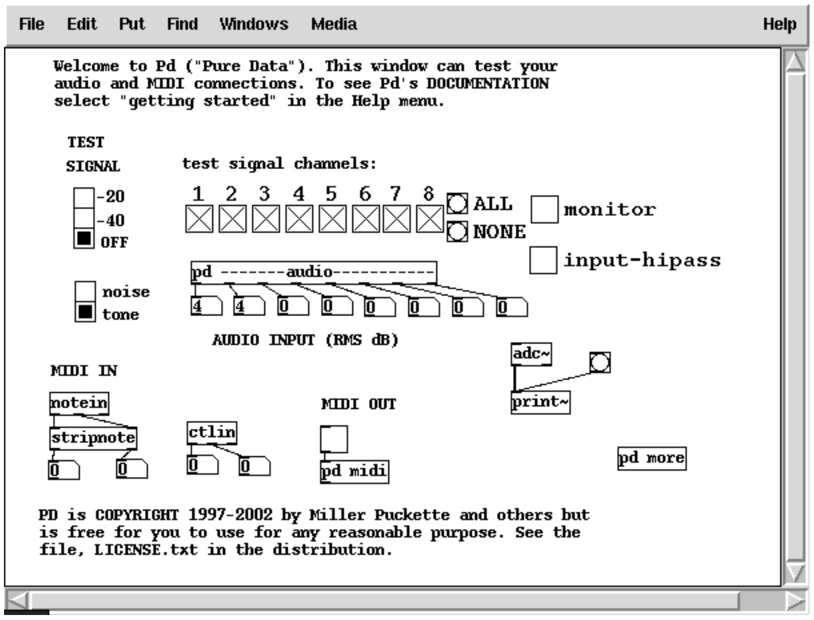


```
iemlib1 (R-1.15) library loaded!
iemlib2 (R-1.15) library loaded!
iem_mp3 (R-1.15) library loaded!
iem_t3_lib (R-1.15) library loaded!
```

fig 2.1: Pure Data console

Testing Pure Data

The first thing to do is turn on the audio and test it. Start by entering the `Media` menu on the top bar and select `Audio ON` (or either check the `compute audio` box in the console window, or press `CTRL+/` on the keyboard.) From the `Media`→`Test-Audio-and-MIDI` menu, turn on the test signal. You should hear a clear tone through your speakers, quiet when set to -40.0dB and much louder when set to -20.0dB . When you are satisfied that Pure Data is making sound close the test window and continue reading. If you don't hear a sound you may need to choose the correct audio settings for your machine. The audio settings summary will look like that shown in Fig. 2.3. Choices available might be Jack, ASIO, OSS, ALSA or the name of a specific device you have installed as a sound card. Most times the default settings will work. If you are using Jack (recommended), then check that Jack audio is running with `qjackctl` on

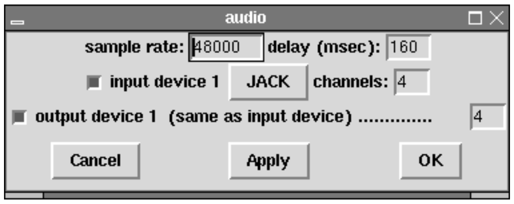


Welcome to Pd ("Pure Data"). This window can test your audio and MIDI connections. To see Pd's DOCUMENTATION select "getting started" in the Help menu.

PD is COPYRIGHT 1997-2002 by Miller Puckette and others but is free for you to use for any reasonable purpose. See the file, LICENSE.txt in the distribution.

fig 2.2: Test signal

Linux or `jack-pilot` on MacOSX. Sample rate is automatically taken from the soundcard.



sample rate: 48000 delay (msec): 160

input device 1 JACK channels: 4

output device 1 (same as input device) 4

fig 2.3: Audio settings pane.

SECTION 2.2

How does Pure Data work?

Pure Data uses a kind of programming called *dataflow*, because the data flows along connections and through objects which process it. The output of one process feeds into the input of another and there may be many steps in the flow.

Objects

Here is a box $\boxed{\quad}$. A musical box, wound up and ready to play. We call these boxes *objects*. Stuff goes in, stuff comes out. For it to pass into, or out of them, objects must have *inlets* or *outlets*. Inlets are at the top of an object box, outlets are at the bottom. Here is an object that has two inlets and one outlet: $\boxed{\quad}$. They are shown by small "tabs" on the edge of the object box. Objects contain processes or procedures which change the things appearing at their inlets and then send the results to one or more outlets. Each object performs some simple function and has a name appearing in its box that identifies what it does. There are two kinds of object, *intrinsics* which are part of the core Pd program, and *externals* which are separate files containing add-ons to the core functions. Collections of externals are called libraries and can be added to extend the functionality of Pd. Most of the time you will neither know nor care whether an object is intrinsic or external. In this book and elsewhere the words *process*, *function* and *unit* are all occasionally used to refer to the object boxes in Pd.

Connections

The connections between objects are sometimes called *cords* or *wires*. They are drawn in a straight line between the outlet of one object and the inlet of another. It is okay for them to cross, but you should try to avoid this since it makes the patch diagram harder to read. At present there are two degrees of thickness for cords. Thin ones carry message data and fatter ones carry audio signals. *Max/MSP*$^{\text{TM}}$ and probably future versions of Pd will offer different colours to indicate the data types carried by wires.

Data

The "stuff" being processed comes in several flavours, video frames, sound signals and messages. In this book we will only be concerned with sounds and messages. Objects give clues about what kind of data they process by their name. For example, an object that adds together two sound signals looks like $\boxed{+\sim}$. The + means this is an addition object, and the $\sim$ (tilde character) means it object operates on signals. Objects without the tilde are used to process messages, which we shall concentrate on before studying audio signal processing.

Patches

A collection of objects wired together is a *program* or *patch*. For historical reasons the words program and patch[2] are used to mean the same thing in sound synthesis. Patches are an older way of describing a synthesiser built from modular units connected together with patch cords. Because inlets and outlets are at the top and bottom of objects the data flow is generally down the patch. Some objects have more than one inlet or more than one outlet, so signals and messages can be a function of many others and may in turn generate multiple

[2]A different meaning of patch to the one programmers use to describe changes made to a program to removes bugs

new data streams. To construct a program we place processing objects onto an empty area called a *canvas*, then connect them together with wires representing pathways for data to flow along. On each step of a Pure Data program any new input data is fed into objects, triggering them to compute a result. This result is fed into the next connected object and so on until the entire chain of objects, starting with the first and ending with the last have all been computed. The program then proceeds to the next step, which is to do the same thing all over again, forever. Each object maintains a state which persists throughout the execution of the program but may change on each step. Message processing objects sit idle until they receive some data rather than constantly processing an empty stream, so we say Pure Data is an *event driven system*. Audio processing objects are always running, unless you explicitly tell them to switch off.

A deeper look at Pd

Before moving on to make some patches consider a quick aside about how Pd actually interprets its patches and how it works in a wider context. A patch, or dataflow graph, is navigated by the interpreter to decide when to compute certain operations. This *traversal* is *right to left* and *depth first*, which is a computer science way of saying it looks a ahead and tries to go as deep as it can before moving on to anything higher and moves from right to left at any branches. This is another way of saying it wants to know what depends on what before deciding to calculate anything. Although we think of data flowing down the graph the nodes in Fig. 2.4 are numbered to show how Pd really thinks about things. Most of the time this isn't very important unless you have to debug a subtle error.

Pure Data software architecture

Pure Data actually consists of more than one program. The main part called **pd** performs all the real work and is the interpreter, scheduler and audio engine. A separate program is usually launched whenever you start the main engine which is called the **pd-gui**. This is the part you will interact with when building Pure Data programs. It creates files to be read by **pd** and automatically passes them to the engine. There is a third program called the **pd-watchdog** which runs as a completely separate process. The job of the watchdog is to keep an eye on the execution of programs by the engine and try to gracefully halt the program if it runs into serious trouble or exceeds available CPU resources. The context of the **pd** program is shown in Fig. 2.5 in terms of other files and devices.

Your first patch

Let's now begin to create a Pd patch as an introductory exercise. We will create some objects and wire them together as a way to explore the interface.

Creating a canvas

A *canvas* is the name for the sheet or window on which you place objects. You can resize a canvas to make it as big as you like. When it is smaller than the patch it contains, horizontal and vertical scrollbars will allow you to change the

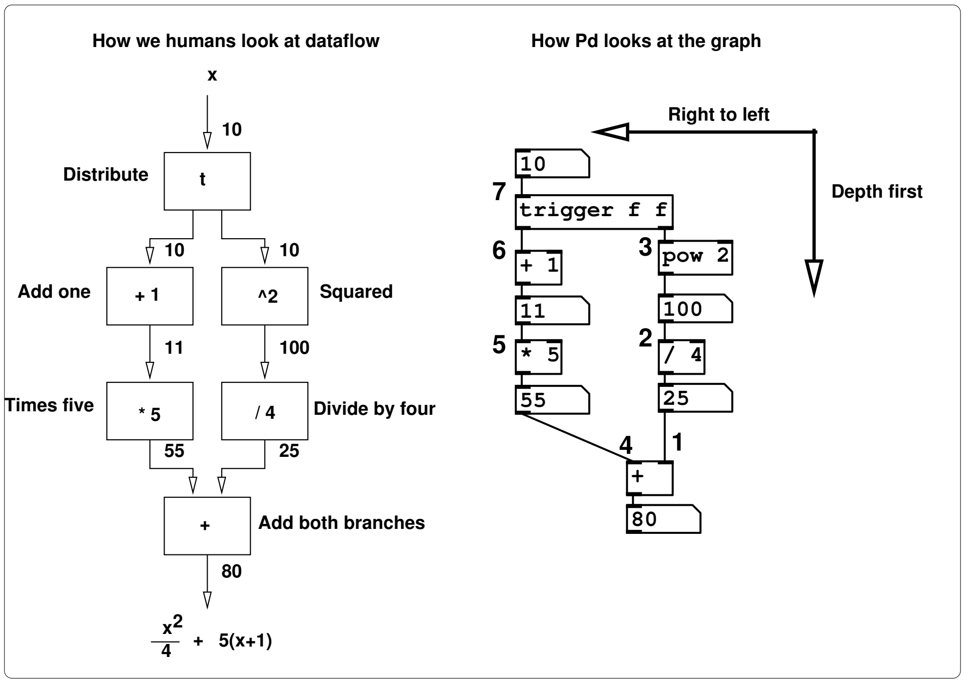


fig 2.4: Dataflow computation

area displayed. When you save a canvas its size and position on the desktop are stored. From the console menu select `File`→`New` or type `CTRL+n` at the keyboard. A new blank canvas will appear on your desktop.

New object placement

To place an object on the canvas select `Put`→`Object` from the menu or use `CTRL+1` on the keyboard. An active, dotted box will appear. Move it somewhere on the canvas using the mouse and click to fix it in place. You can now type the name of the new object, so type the multiply character `*` into the box. When you have finished typing click anywhere on the blank canvas to complete the operation. When Pure Data recognises the object name you give, it immediately changes the object box boundary to a solid line and adds a number of inlets and outlets. You should see a `*` on the canvas now.

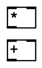


fig 2.6: Objects on a canvas

Pure Data searches the paths it knows for objects, which includes the current working directory. If it doesn't recognise an object because it can't find a definition anywhere the boundary of the object box remains dotted. Try creating another object and typing some nonsense into it, the boundary will stay dotted and no inlets or outlets will be assigned. To delete the object place the mouse cursor close to it, click and hold in order to draw

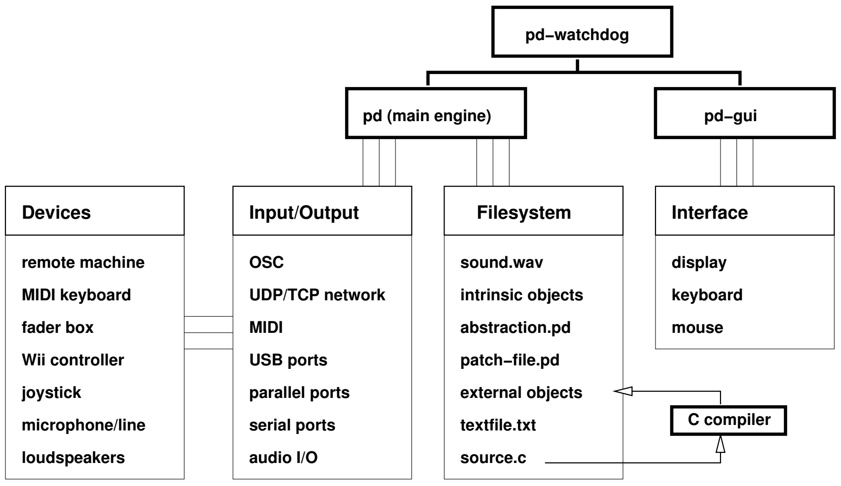


fig 2.5: Pure Data software architecture

a selection box around it, then hit `delete` on the keyboard. Create another object beneath the last one with an addition symbol so your canvas looks like Fig. 2.6

Edit mode and wiring

When you create a new object from the menu Pd automatically enters edit mode, so if you just completed the instructions above you should currently be in edit mode. In this mode you can make connections between objects, or delete objects and connections.

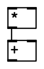


fig 2.7: Wiring objects

Hovering over an outlet will change the mouse cursor to a new “wiring tool”. If you click and hold the mouse when the tool is active you will be able to drag a connection away from the object. Hovering over a compatible inlet while in this state will allow you to release the mouse and make a new connection. Connect together the two objects you made so that your canvas looks like Fig. 2.7. If you want to delete a connection it’s easy, click on the connection to select it and then hit the `delete` key. When in edit mode you can move any object to another place by clicking over it and dragging with the mouse. Any connections already made to the object will follow along. You can pick up and move more than one object if you draw a selection box around them first.

Initial parameters

Most objects can take some initial parameters or *arguments*, but these aren’t always required. They can be created without any if you are going to pass data via the inlets as the patch is running. The `+` object can be written as `+ 3` to create an object which always adds 3 to its input. Uninitialised values generally resort to zero so the default behaviour of + would be to add 0 to its input, which is the same as doing nothing. Contrast this to the default behaviour of * which always gives zero.

Modifying objects

You can also change the contents of any object box to alter the name and function, or to add parameters.

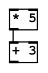


fig 2.8: Changing objects

In Fig. 2.8 the objects have been changed to give them initial parameters. The multiply object is given a parameter of 5, which means it multiplies its input by 5 no matter what comes in. If the input is 4 then the output will be 20. To change the contents of an object click on the middle of the box where the name is and type the new text. Alternatively click once, and then again at the end of the text to append new stuff, such as adding 5 and 3 to the objects shown in Fig. 2.8

Number input and output

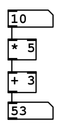


fig 2.9: Number boxes

One of the easiest ways to create and view numerical data is to use number boxes. These can act as input devices to generate numbers, or as displays to show you the data on a wire. Create one by choosing `Put`→`Number` from the canvas menu, or use `CTRL+3`, and place it above the * object. Wire it to the left inlet. Place another below the + object and wire the object outlet to the top of the number box as shown in Fig. 2.9.

Toggling edit mode

Pressing `CTRL+E` on the keyboard will also enter edit mode. This key combination toggles modes, so hitting `CTRL+E` again exits edit mode. Exit edit mode now by hitting `CTRL+E` or selecting `Edit`→`Edit mode` from the canvas menu. The mouse cursor will change and you will no longer be able to move or modify object boxes. However, in this mode you can operate the patch components such as buttons and sliders normally. Place the mouse in the top number box, click and hold and move it upwards. This input number value will change, and it will send messages to the objects below it. You will see the second number box change too as the patch computes the equation $y = 5x + 3$. To re-enter edit mode hit `CTRL+E` again or place a new object.

More edit operations

Other familiar editing operations are available while in edit mode. You can cut or copy objects to a buffer or paste them back into the canvas, or to another canvas opened with the same instance of Pd. Take care with pasting objects in the buffer because they will appear directly on top of the last object copied. To select a group of objects you can drag a box around them with the mouse.

Holding `SHIFT` while selecting allows multiple separate objects to be added to the buffer.

- `CTRL+A` Select all objects on canvas.
- `CTRL+D` Duplicate the selection.
- `CTRL+C` Copy the selection.
- `CTRL+V` Paste the selection.
- `CTRL+X` Cut the selection.
- `SHIFT` Select multiple objects.

Duplicating a group of objects will also duplicate any connections between them. You may modify an object once created and wired up without having it disconnect so long as the new one is compatible the existing inlets and outlets, for example replacing + with -. Clicking on the object text will allow you to retype the name and, if valid, the old object is deleted and its replacement remains connected as before.

Patch files

Pd files are regular text files in which patches are stored. Their names always end with a **.pd** file extension. Each consists of a *netlist* which is a collection of object definitions and connections between them. The file format is terse and difficult to understand, which is why we use the GUI for editing. Often there is a one to one correspondence between a patch, a single canvas, and a file, but you can work using multiple files if you like because all canvases opened by the same instance of Pd can communicate via global variables or through send and receive objects. Patch files shouldn't really be modified in a text editor unless you are an expert Pure Data user, though a plaintext format is useful because you can do things like search for and replace all occurrences of an object. To save the current canvas into a file select `File→Save` from the menu or use the keyboard shortcut `CTRL+s`. If you have not saved the file previously a dialogue panel will open to let you choose a location and file name. This would be a good time to create a folder for your Pd patches somewhere convenient. Loading a patch, as you would expect, is achieved with `File→Open` or `CTRL+o`.

SECTION 2.3

Message data and GUI boxes

We will briefly tour the basic data types that Pd uses along with GUI objects that can display or generate that data for us. The message data itself should not be confused with the objects that can be used to display or input it, so we distinguish messages from boxes. A *message* is an event, or a piece of data that gets sent between two objects. It is invisible as it travels down the wires, unless we print it or view it in some other way like with the number boxes above. A message can be very short, only one number or character, or very long, perhaps holding an entire musical score or synthesiser parameter set. They can be floating point numbers, lists, symbols, or pointers which are references to other types like datastructures. Messages happen in *logical time*, which means that they aren't synchronised to any real timebase. Pd processes them as fast as it can, so when you change the input number box, the output number box changes instantly. Let's look at some other message types we'll encounter while building patches to create sound. All GUI objects can be placed on a canvas using the `Put` menu or using keyboard shortcuts `CTRL+1` through `CTRL+8`, and all have *properties* which you can access by clicking them while in edit mode and selecting the `properties` pop-up menu item. Properties include things like colour, ranges, labels and size and are set per instance.

Selectors

With the exception of a bang message, all other message types carry an invisible *selector*, which is a symbol at the head of the message. This describes the "type" of the remaining message, whether it represents a symbol, number, pointer or list. Object boxes and GUI components are only able to handle appropriate messages. When a message arrives at an inlet the object looks at the selector and searches to see if it knows of an appropriate *method* to deal with it. An error results when an incompatible data type arrives at an inlet, so for example, if you supply a symbol type message to a `delay` object it will complain...

```
error: delay: no method for 'symbol'
```

Bang message

This is the most fundamental, and smallest message. It just means "compute something". Bangs cause most objects to output their current value or advance to their next state. Other messages have an implicit bang so they don't need to be followed with a bang to make them work. A bang has no value, it is just a bang.

Bang box

A bang box looks like this, ◻ and sends and receives a bang message. It briefly changes colour, like this ◼, whenever it is clicked or upon receipt of a bang message to show you one has been sent or received. These may be used as buttons to initiate actions or as indicators to show events.

Float messages

Floats are another name for numbers. As well as regular (integer) numbers like 1, 2, 3 and negative numbers like -10 we need numbers with decimal points like -198753.2 or 10.576 to accurately represent numerical data. These are called floating point numbers, because of the way computers represent the decimal point position. If you understand some computer science then it's worth noting that there are no integers in Pd, everything is a float, even if it appears to be an integer, so 1 is really 1.0000000. Current versions of Pd use a 32 bit float representation, so they are between -8388608 and 8388608.

Number box

For float numbers we have already met the number box, which is a dual purpose GUI element. Its function is to either display a number, or allow you to input one. A bevelled top right corner like this 0 denotes that this object is a number box. Numbers received on the inlet are displayed and passed directly to the outlet. To input a number click and hold the mouse over the value field and move the mouse up or down. You can also type in numbers. Click on a number box, type the number and hit `RETURN`. Number boxes are a compact replacement for faders. By default it will display up to five digits including a sign if negative, -9999 to 99999, but you can change this by editing its properties. Holding `SHIFT` while moving the mouse allows a finer degree of control. It is also possible to set an upper and lower limit from the `properties` dialog.

Toggle box

Another object that works with floats is a toggle box. Like a checkbox on any standard GUI or web form, this has only two states, on or off. When clicked a cross appears in the box like ☒ and it sends out a number 1, clicking again causes it to send out a number 0 and removes the cross so it looks like this ☐. It also has an inlet which sets the value, so it can be used to display a binary state. Sending a bang to the inlet of a toggle box does not cause the current value to be output, instead it flips the toggle to the opposite state and outputs this value. Editing `properties` also allows you to send numbers other than 1 for the active state.

Sliders and other numerical GUI elements

GUI elements for horizontal and vertical sliders can be used as input and display elements. Their default range is 0 to 127, nice for MIDI controllers, but like all other GUI objects this can be changed in their `properties` window. Unlike those found in some other GUI systems, Pd sliders do not have a step value. Shown in Fig. 2.10 are some GUI objects at their standard sizes. They can be

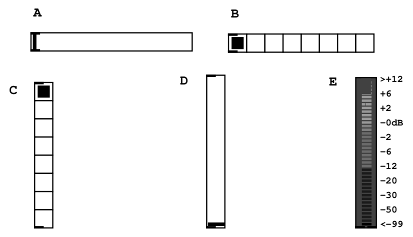


fig 2.10: GUI Objects A: Horizontal slider B: Horizontal radio box C: Vertical radio box D: Vertical slider E: VU meter

ornamented with labels or created in any colour. Resizing the slider to make it bigger will increase the step resolution. A radio box provides a set of mutually exclusive buttons which output a number starting at zero. Again, they work equally well as indicators or input elements. A better way to visually display an audio level is to use a VU meter. This is set up to indicate decibels, so has a rather strange scale from -99.0 to $+12.0$. Audio signals that range from -1.0

to +1.0 must first be scaled using the appropriate object. The VU is one of the few GUI elements that only acts as a display.

General messages

Floats and bangs are types of message, but messages can be more general. Other message types can be created by prepending a *selector* that gives them special meanings. For example, to construct lists we can prepend a *list* selector to a set of other types.

Message box

These are visual containers for user definable messages. They can be used to input or store a message. The right edge of a message box is curved inwards like this , and it always has only one inlet and one outlet. They behave as GUI elements, so when you click a message box it sends its contents to the outlet. This action can also be triggered if the message box receives a bang message on its inlet. Message boxes do some clever thinking for us. If we store something like `5.0` it knows that is a float and outputs a float type, but if we create `a message with text` then it will send out a list of symbols, so it is type aware which saves us having to say things like "float 1.0" as we would in C programs. It can also abbreviate floating point numbers like 1.0 to 1, which saves time when inputting integer values, but it knows that they are really floats.

Symbolic messages

A *symbol* generally is a word or some text. A symbol can represent anything, it is the most basic textural message in Pure Data. Technically a symbol in Pd can contain any printable or non-printable character. But most of the time you will only encounter symbols made out of letters, numbers and some interpunctuation characters like dash, dot or underscore. The Pd editor does some automatic conversions: words that can also be interpreted as a number (like 3.141 or $1e+20$) are converted to a float internally (but +20 still is a symbol!). Whitespace is used by the editor to separate symbols from each other, so you cannot type a symbol including a space character into a message box. To generate symbols with backslash-escaped whitespace or other special characters inside use the `makefilename` symbol maker object. The `openpanel` file dialog object preserves and escapes spaces and other special characters in filenames, too. Valid symbols are *badger*, *sound_2*, or *all_your_base* but not *hello there* (which is two symbols), or *20* (which will be interpreted as a float, 20.0).

Symbol box

For displaying or inputting text you may use a `symbol` box. Click on the display field and type any text that is a valid symbol and then hit `ENTER/RETURN`. This will send a symbol message to the outlet of the box. Likewise, if a symbol message is received at the inlet it will be displayed as text. Sending a bang message to a symbol box makes it output any symbol it already contains.

Lists

A list is an ordered collection of any things, floats, symbols or pointers that are treated as one. Lists of floats might be used for building melody sequences or setting the time values for an envelope generator. Lists of symbols can be used to represent text data from a file or keyboard input. Most of the time we will be interested in lists of numbers. A list like {*2 127 3.14159 12*} has four elements, the first element is 2.0 and the last is 12.0. Internally, Pure Data recognises a list because it has a *list selector* at the start, so it treats all following parts of the message as ordered list elements. When a list is sent as a message all its elements are sent at once. A list selector is attached to the beginning of the message to determine its type. The selector is the word "list", which has a special meaning to Pd. Lists may be of mixed types like {*5 6 pick up sticks*}, which has two floats and three symbols. When a list message contains only one item which is a float it is automatically changed (cast) back to a float. Lists can be created in several ways, by using a message box, or by using pack, which we will meet later, to pack data elements into a list.

Pointers

As in other programming languages, a *pointer* is the address of some other piece of data. We can use them to build more complex datastructures, such as a pointer to a list of pointers to lists of floats and symbols. Special objects exist for creating and dereferencing pointers, but since they are an advanced topic we will not explore them further in this book.

Tables, arrays and graphs

A *table* is sometimes used interchangeably with an *array* to mean a two dimensional data structure. An array is one of the few invisible objects. Once declared it just exists in memory. To see it, a separate *graph* like that shown in Fig. 2.11 allows us to view its contents.

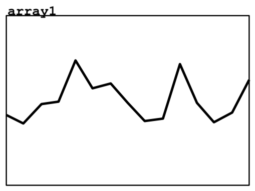
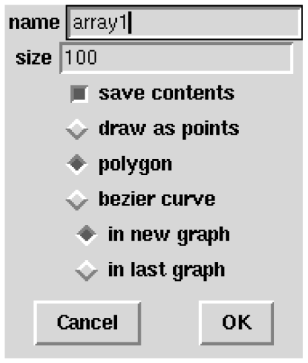


fig 2.11: An array.

Graphs have the wonderful property that they are also GUI elements. You can draw data directly into a graph using the mouse and it will modify the array it is attached to. You can see a graph of **array1** in Fig. 2.11 that has been drawn by hand. Similarly, if the data in an array changes and it's attached to a visible graph then the graph will show the data as it updates. This is perfect for drawing detailed envelopes or making an oscilloscope display of rapidly changing signals.



name array1

size 100

■ save contents

◇ draw as points

◆ polygon

◇ bezier curve

◆ in new graph

◇ in last graph

fig 2.12: Create array.

To create a new array select Put→Array from the menu and complete the dialog box to set up its name, size and display characteristics. On the canvas a graph will appear showing an array with all its values initialised to zero. The Y-axis range is −1.0 to +1.0 by default, so the data line will be in the centre. If the `save contents` box is checked then the array data will be saved along with the patch file. Be aware that long sound files stored in arrays will make large patch files when saved this way. Three draw styles are available, points, polygon and Bezier to show the data with varying degrees of smoothing. It is possible to use the same graph to display more than one array, which is very useful when you wish to see the relationship between two or more sets of data. To get this behaviour use the `in last graph` option when creating an array.

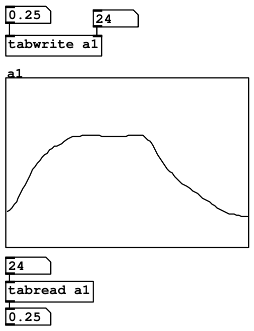


fig 2.13: Accessing an array.

Data is written into or read from a table by an index number which refers to a position within it. The index is a whole number. To read and write arrays several kinds of accessor object are available. The `tabread` and `tabwrite` objects allow you to communicate with arrays using messages. Later we will meet `tabread4~` and `tabwrite~` objects that can read and write audio signals. The array **a1** shown in Fig. 2.13 is written to by the `tabwrite` object above it, which specifies the target array name as a parameter. The right inlet sets the index and the left one sets the value. Below it a `tabread` object takes the index on its inlet and returns the current value.

SECTION 2.4 Getting help with Pure Data

At **http://puredata.hurleur.com/** there is an active, friendly forum, and the mailing list can be subscribed to at **pd-list@iem.at**

Exercises

Exercise 1

On Linux, type `pd --help` at the console to see the available startup options. On Windows or MacOSX read the help documentation that comes with your downloaded distribution.

Exercise 2

Use the `Help` menu, select `browse help` and read through some built in documentation pages. Be familiar with the `control examples` and `audio examples` sections.

Exercise 3

Visit the online `pdwiki` at http://puredata.org to look at the enormous range of objects available in `pd-extended`.

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Arduino I/O boards http://www.arduino.cc/

CHAPTER 3
Using Pure Data

SECTION 3.1
Basic objects and principles of operation

Now we are familiar with the basics of Pd let's look at some essential objects and rules for connecting them together. There are about 20 message objects you should try to learn by heart because almost everything else is built from them.

Hot and cold inlets

Most objects operating on messages have a "hot" inlet and (optionally) one or more "cold" inlets. Messages received at the hot inlet, usually the leftmost one, will cause computation to happen and output to be generated. Messages on a cold inlet will update the internal value of an object but not cause it to output the result yet. This seems strange at first, like a bug. The reason is so that we can order evaluation. This means waiting for sub-parts of a program to finish in the right order before proceeding to the next step. From maths you know that brackets describe the order of a calculation. The result of $4 \times 10 - 3$ is not the same as $4 \times (10 - 3)$, we need to calculate the parenthesised parts first. A Pd program works the same way, you need to wait for the results from certain parts before moving on.

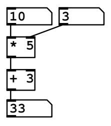


fig 3.1: Hot and cold inlets

In Fig. 3.1 a new number box is added to right inlet of *. This new value represents a constant multiplier k so we can compute $y = kx + 3$. It overrides the 5 given as an initial parameter when changed. In Fig. 3.1 it's set to 3 so we have $y = 3x + 3$. Experiment setting it to another value and then changing the left number box. Notice that changes to the right number box don't immediately effect the output, because it connects to the cold inlet of *, but changes to the left number box cause the output to change, because it is connected to the hot inlet of *.

Bad evaluation order

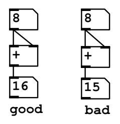


fig 3.2: Bad ordering

A problem arises when messages fan out from a single outlet into other operations. Look at the two patches in Fig. 3.2. Can you tell the difference? It is impossible to tell just by looking that one is a working patch and the other contains a nasty error. Each is an attempt to double the value of a number by connecting it to both sides of a `+`. When connections are made this way the behaviour is undefined, but usually happens in the order the connections were made. The first one works because the right (cold) inlet was connected before the left (hot) one. In the second patch the arriving number is added to the *last* number received because the hot inlet is addressed first. Try making these patches by connecting the inlets to `+` in a different order. If you accidentally create errors this way they are hard to debug.

Trigger objects

A trigger is an object that splits a message up into parts and sends them over several outlets in order. It solves the evaluation order problem by making the order explicit.

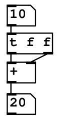


fig 3.3: Ordering with trigger

The order of output is right to left, so a `trigger bang float` object outputs a float on the right outlet first, then a bang on the left one. This can be abbreviated as `t b f`. Proper use of triggers ensures correct operation of units further down the connection graph. The arguments to a trigger may be **s** for symbol, **f** for float, **b** for bang, **p** for pointers and **a** for any. The "any" type will pass lists and pointers too. The patch in Fig. 3.3 always works correctly, whatever order you connect to the `+` inlets. The float from the right outlet of `t f f` is always sent to the cold inlet of `+` first, and the left one to the hot inlet afterwards.

Making cold inlets hot

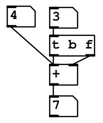


fig 3.4: Warming an inlet

An immediate use for our new knowledge of triggers is to make an arithmetic operator like `+` respond to either of its inlets immediately. Make the patch shown in Fig. 3.4 and try changing the number boxes. When the left one is changed it sends a float number message to the left (hot) inlet which updates the output as usual. But now, when you change the right number box it is split by `t b f` into two messages, a float which is sent to the cold (right) inlet of `+`, and a bang, which is sent to the hot inlet immediately afterwards. When it receives a bang on its hot inlet `+` computes the sum of the two numbers last seen on its inlets, which gives the right result.

Float objects

The object `f` is very common. A shorthand for `float`, which you can also use if you like to make things clearer, it holds the value of a single floating point number. You might like to think of it as a variable, a temporary place to store a number. There are two inlets on `f`, the rightmost one will set the value of the object, and the leftmost one will both set the value and/or output it depending on what message it receives. If it receives a bang message it will just output whatever value is currently stored, but if the message is a float it will override the currently stored value with a new float and immediately output that. This gives us a way to both set and query the object contents.

Int objects

Although we have noted that integers don't really exist in Pd, not in a way that a programmer would understand, whole numbers certainly do. int stores a float as if it were an integer in that it provides a rounding (truncation) function of any extra decimal places. Thus 1.6789 becomes 1.0000, equal to 1, when passed to int.

Symbol and list objects

As for numbers there are likewise object boxes to store lists and symbols in a temporary location. Both work just like their numerical counterparts. A list can be given to the right inlet of list and recalled by banging the left inlet. Similarly symbol can store a single symbol until it is needed.

Merging message connections

When several message connections are all connected to the same inlet that's fine. The object will process each of them as they arrive, though it's up to you to ensure that they arrive in the right order to do what you expect. Be aware of race hazards when the sequence is important.

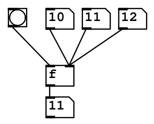


fig 3.5: Messages to same inlet

Messages arriving from different sources at the same hot inlet have no effect on each another, they remain separate and are simply interleaved in the order they arrive, each producing output. But be mindful that where several connections are made to a cold inlet only the last one to arrive will be relevant. Each of the number boxes in Fig. 3.5 connects to the same cold inlet of the float box f and a bang button to the hot inlet. Whenever the bang button is pressed the output will be whatever is currently stored in f, which will be the last number box changed. Which number box was updated last in Fig. 3.5? It was the middle one with a value of 11.

SECTION 3.2

Working with time and events

With our simple knowledge of objects we can now begin making patches that work on functions of time, the basis of all sound and music.

Metronome

Perhaps the most important primitive operation is to get a beat or timebase. To get a regular series of bang events metro provides a clock. Tempo is given as a period in milliseconds rather than beats per minute (as is usual with most music programs).

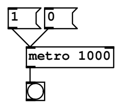


fig 3.6: Metronome

The left inlet toggles the metronome on and off when it receives a 1 or 0, while the right one allows you to set the period. Periods that are fractions of a millisecond are allowed. The metro emits a bang as soon as it is switched on and the following bang occurs after the time period. In Fig. 3.6 the time period is 1000ms, (equal to 1 second). The bang button here is used as an indicator. As soon as you click the message box to send 1 to metro it begins sending out bangs which make the bang button flash once per second, until you send a 0 message to turn it off.

A counter timebase

We could use the metronome to trigger a sound repeatedly, like a steady drum beat, but on their own a series of bang events aren't much use. Although they are separated in time we cannot keep track of time this way because bang messages contain no information.

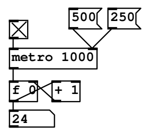


fig 3.7: Counter

In Fig. 3.7 we see the metronome again. This time the messages to start and stop it have been conveniently replaced by a toggle switch. I have also added two new messages which can change the period and thus make the metronome faster or slower. The interesting part is just below the metronome. A float box receives bang messages on its hot inlet. Its initial value is 0 so upon receiving the first bang message it outputs a float number 0 which the number box then displays. Were it not for the + 1 object the patch would continue outputting 0 once per beat forever. However, look closely at the wiring of these two objects, f and + 1 are connected to form an *incrementor* or *counter*. Each time f recieves a bang it ouputs the number currently stored to + 1 which adds 1 to it. This is fed back into the cold inlet of f which updates its value, now 1. The next time a bang arrives 1 is output, which goes round again, through + 1 and becomes 2. This repeats as long as bang messages arrive, each time the output increases by 1. If you start the metronome in Fig. 3.7 you will see the number box slowly counting up, once per second. Clicking the message boxes to change the period will make it count up faster with a 500ms delay between beats (twice per second), or still faster at 4 times per second (250ms period).

Time objects

Three related objects help us manipulate time in the message domain. timer accurately measures the interval between receiving two bang messages, the first on its left inlet and the second on its right inlet. It is shown on the left of Fig. 3.8.

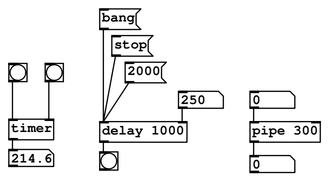


fig 3.8: Time objects

Clicking the first bang button will reset and start `timer` and then hitting the second one will output the time elapsed (in ms). Notice that `timer` is unusual, it's one of the few objects where the right inlet behaves as the hot control. `delay` shown in the middle of Fig. 3.8 will output a single bang message a certain time period after receiving a bang on its left inlet. This interval is set by its first argument or right inlet, or by the value of a float arriving at its left inlet, so there are three ways of setting the time delay. If a new bang arrives any pending one is cancelled and a new delay is initiated. If a `stop` message arrives then `delay` is reset and all pending events are cancelled. Sometimes we want to delay a stream of number messages by a fixed amount, which is where `pipe` comes in. This allocates a memory buffer that moves messages from its inlet to its outlet, taking a time set by its first argument or second inlet. If you change the top number box of the right patch in Fig. 3.8 you will see the lower number box follow it, but lagging behind by 300ms.

Select

This object outputs a bang on one of its outlets matching something in its argument list. For example `select 2 4 6` will output a bang on its second outlet if it receives a number 4, or on its third outlet when a number 6 arrives. Messages that do not match any argument are passed through to the rightmost outlet.

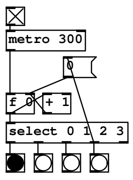


fig 3.9: Simple sequencer

This makes it rather easy to begin making simple sequences. The patch in Fig. 3.9 cycles around four steps blinking each bang button in turn. It is a metronome running with a 300ms period and a counter. On the first step the counter holds 0, and when this is output to `select` it sends a bang to its first outlet which matches 0. As the counter increments, successive outlets of `select` produce a bang, until the fourth one is reached. When this happens a message containing 0 is triggered which feeds into the cold inlet of `f` resetting the counter to 0.

SECTION 3.3

Data flow control

In this section are a few common objects used to control the flow of data around patches. As you have just seen `select` can send bang messages along a choice of connections, so it gives us a kind of selective flow.

Route

Route behaves in a similar fashion to select, only it operates on lists. If the first element of a list matches an argument the remainder of the list is passed to the corresponding outlet.

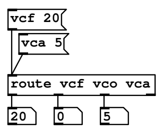


fig 3.10: Routing values

So, `route badger mushroom snake` will send 20.0 to its third outlet when it receives the message {*snake 20*}. Non matching lists are passed unchanged to the rightmost outlet. Arguments can be numbers or symbols, but we tend to use symbols because a combination of `route` with lists is a great way to give parameters names so we don't forget what they are for. We have a few named values in Fig. 3.10 for synthesiser controls. Each message box contains a two element list, a name-value pair. When `route` encounters one that matches one of its arguments it sends it to the correct number box.

Moses

A "stream splitter" which sends numbers below a threshold to its left outlet, and numbers greater than or equal to the threshold to the right outlet. The threshold is set by the first argument or a value appearing on the right inlet. `moses 20` splits any incoming numbers at 20.0

Spigot

This is a switch that can control any stream of messages including lists and symbols. A zero on the right inlet of `spigot` stops any messages on the left inlet passing to the outlet. Any non-zero number turns the spigot on.

Swap

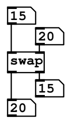


fig 3.11: Swapping values

It might look like a very trivial thing to do, and you may ask - why not just cross two wires? In fact `swap` is really useful object. It just exchanges the two values on its inlets and passes them to its outlets, but it can take an argument so it always exchanges a number with a constant. It's useful when this constant is 1 as shown later for calculating complement $1 - x$ and inverse $1/x$ of a number, or where it is 100 for calculating values as a percent.

Change

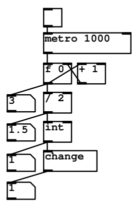


fig 3.12: Pass values that change

This is useful if we have a stream of numbers, perhaps from a physical controller like a joystick that is polled at regular intervals, but we only want to know values when they change. It is frequently seen preceded by `int` to denoise a jittery signal or when dividing timebases. In Fig. 3.12 we see a counter that has been stopped after reaching 3. The components below it are designed to divide the timebase in half. That is to say, for a sequence {*1, 2, 3, 4, 5, 6 ...*} we will get {*1, 2, 3 ...*}. There should be half as many numbers in the output during the same time interval. In other words the output changes half as often as the input. Since the counter has just passed 3 the output of `/` is 1.5 and `int` truncates this to 1. But this is the second time we have seen 1 appear, since the same number was sent when the input was 2. Without using `change` we would get {*1, 1, 2, 2, 3, 3 ...*} as output.

Send and receive objects

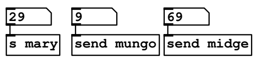


fig 3.13: Sends

Very useful for when patches get too visually dense, or when you are working with patches spread across many canvases. `send` and `receive` objects, abbreviated as `s` and `r` work as named pairs. Anything that goes into the send unit is transmitted by an invisible wire and appears immediately on the receiver, so whatever goes into `send bob` reappears at `receive bob`.

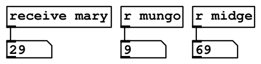


fig 3.14: Receives

Matching sends and receives have global names by default and can exist in different canvases loaded at the same time. So if the `receive` objects in Fig. 3.14 are in a different patch they will still pick up the send values from Fig. 3.13. The relationship is one to many, so only one send can have a particular name but can be picked up by multiple `receive` objects with the same name. In the latest versions of Pd the destination is dynamic and can be changed by a message on the right inlet.

Broadcast messages

As we have just seen there is an "invisible" environment through which messages may travel as well as through wires. A message box containing a message that begins with a semicolon is *broadcast* and Pd will route it to any destination that matches the first symbol. This way, activating the message box `; foo 20` is the same as sending a float message with a value of 20 to the object `s foo`.

Special message destinations

This method can be used to address arrays with special commands, to talk to GUI elements that have a defined *receive symbol* or as an alternative way to talk to `receive` objects. If you want to change the size of arrays dynamically they recognise a special *resize* message. There is also a special destination (which always exists) called `pd` which is the audio engine. It can act on broadcast messages like `; pd dsp 1` to turn on the audio computation from a patch. Some examples are shown in Fig. 3.15

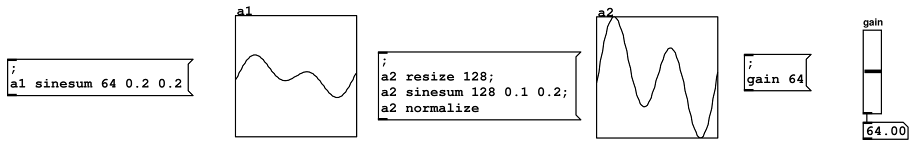


fig 3.15: Special message broadcasts

Message sequences

Several messages can be stored in the same message-box as a sequence if separated by commas, so `2, 3, 4, 5` is a message-box that will send four values one after another when clicked or banged. This happens instantly (in *logical time*). This is often confusing to beginners when comparing sequences to lists. When you send the contents of a message box containing a sequence all the elements are sent in one go, but as separate messages in a stream. Lists on the other hand, which are not separated by commas, also send all the elements at the same time, but as a single list message. Lists and sequences can be mixed, so a message box might contain a sequence of lists.

SECTION 3.4

List objects and operations

Lists can be quite an advanced topic and we could devote an entire chapter to this subject. Pd has all the capabilities of a full programming language like LISP, using only list operations, but like that language all the more complex functions are defined in terms of just a few intrinsic operations and abstractions. The *list-abs* collection by Frank Barknecht and others is available in *pd-extended*. It contains scores of advanced operations like sorting, reversing, inserting, searching and performing conditional operations on every element of a list. Here we will look at a handful of very simple objects and leave it as an exercise to the reader to research the more advanced capabilities of lists for building sequencers and data analysis tools.

Packing and unpacking lists

The usual way to create and disassemble lists is to use `pack` and `unpack`. Arguments are given to each which are type identifiers, so `pack f f f f` is an object that will wrap up four floats given on its inlets into a single list. They should be presented in right to left order so that the hot inlet is filled last. You can also give float values directly as arguments of a `pack` object where you want them to be fixed, so `pack 1 f f 4` is legal, the first and last list elements will be 1 and 4 unless over-ridden by the inlets, and the two middle ones will be variable.

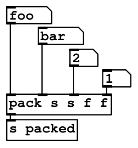


fig 3.16: List packing

Start by changing the right number in Fig. 3.16, then the one to its left, then click on the symbol boxes and type a short string before hitting RETURN. When you enter the last symbol connected to the hot inlet of `pack` you will see the data received by Fig. 3.17 appear in the display boxes after it is unpacked.

The `unpack s s f f` will expect two symbols and two floats and send them to its four outlets. Items are packed and unpacked in the sequence given in the list, but in right to left order. That means the floats from `unpack s s f f` will appear first, starting with the rightmost one, then the two symbols ending on the leftmost one. Of course this happens so quickly you cannot see the ordering, but it makes sense to happen this way so that if you are unpacking data, changing it and re-packing into a list everything occurs in the right order. Note that the types of data in the list must match the arguments of each object. Unless you use the **a** (any) type Pd will complain if you try to pack or unpack a mismatched type.

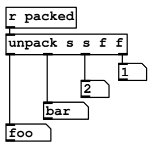


fig 3.17: List unpacking

Substitutions

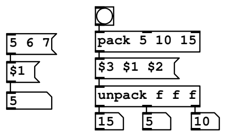


fig 3.18: Dollar substitution.

A message box can also act as a template. When an item in a message box is written $1 it behaves as an empty slot that assumes the value of the first element of a given list. Each of the dollar arguments $1, $2 and so on, are replaced by the corresponding item in the input list. The message box then sends the new message with any slots filled in. List elements can be substituted in multiple positions as seen in Fig. 3.18. The list {*5 10 15*} becomes {*15 5 10*} when put through the substitution `$3 $1 $2`.

Persistence

You will often want to set up a patch so it's in a certain state when loaded. It's possible to tell most GUI objects to output the last value they had when the patch was saved. You can do this by setting the `init` checkbox in the `properties` panel. But what if the data you want to keep comes from another source, like an external MIDI fader board? A useful object is `loadbang` which generates a bang message as soon as the patch loads.

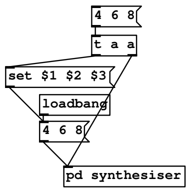


fig 3.19: Persistence using messages

You can use this in combination with a message box to initialise some values. The contents of message boxes are saved and loaded with the patch. When you need to stop working on a project but have it load the last state next time around then list data can be saved in the patch with a message box by using the special `set` prefix. If a message box receives a list prefixed by `set` it will be filled with the list, but not immediately ouput it. The arrangement in Fig. 3.19 is used to keep a 3 element list for `pd synthesiser` in a message box that will be saved with the patch, then generate it to initialise the synthesiser again when the patch is reloaded.

List distribution

An object with 2 or more message inlets will distribute a list of parameters to all inlets using only the first inlet.

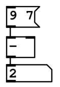


fig 3.20: Distribution

The number of elements in the list must match the number of inlets and their types must be compatible. In Fig. 3.20 a message box contains a list of two numbers, 9 and 7. When a pair of values like this are sent to - with its right inlet unconnected they are spread over the two inlets, in the order they appear, thus $9 - 7 = 2$.

More advanced list operations

To concatenate two lists together we use list append. It takes two lists and creates a new one with the second list attached to the end of the first. If given an argument it will append this to every list it receives. It may be worth knowing that list is an alias for list append. You can choose to type in either in order to make it clearer what you are doing. Very similar is list prepend which does almost the same, but returns a new list with the argument or list at the second inlet concatenated to the beginning. For disassembling lists we can use list split. This takes a list on its left inlet and a number on the right inlet (or as an argument) which indicates the position to split the list. It produces two new lists, one containing elements below the split point appears on the left outlet, and the remainder of the list appears on the right. If the supplied list is shorter than the split number then the entire list is passed unchanged to the right outlet. The list trim object strips off any selector at the start leaving the raw elements.

SECTION 3.5

Input and output

There are plenty of objects in Pd for reading keyboards, mice, system timers, serial ports and USB. There's not enough room in this book to do much more than summarise them, so please refer to the Pd online documentation for your platform. Many of these are available only as external objects, but several are built into Pd core. Some depend on the platform used, for example comport and key are only available on Linux and MacOS. One of the most useful externals available is hid which is the "human interface device". With this you can connect joysticks, game controllers, dance mats, steering wheels, graphics tablets and all kinds of fun things. File IO is available using textfile and qlist objects, objects are available to make database transactions to MySQL, and of course audio file IO is simple using a range of objects like writesf~ and readsf~. MIDI files can be imported and written with similar objects. Network access is available through netsend and netreceive which offer UDP or TCP services. Open Sound Control is available using the external OSC library by Martin Peach or dumpOSC and sendOSC objects. You can even generate or open compressed audio streams using mp3cast~ and similar externals, and you can run code from other languages like python and lua. A popular hardware peripheral for use in combination with

Pd is the Arduino board which gives a number of buffered analog and digital lines, serial and parallel, for robotics and control applications. Nearly all of this is quite beyond the scope of this book. The way you set up your DAW and build your sound design studio is an individual matter, but Pd should not disappoint you when it comes to I/O connectivity. We will now look at a few common input and output channels.

The print object

Where would we be without a `print` object? Not much use for making sound, but vital for debugging patches. Message domain data is dumped to the console so you can see what is going on. You can give it a non-numerical argument which will prefix any output and make it easier to find in a long printout.

MIDI

When working with musical keyboards there are objects to help integrate these devices so you can build patches with traditional synthesiser and sampler behaviours. For sound design this is great for attaching MIDI fader boards to control parameters, and of course musical interface devices like breath controllers and MIDI guitars can be used. Hook up any MIDI source to Pd by activating a MIDI device from the `Media->MIDI` menu (you can check this is working from `Media->Test Audio and MIDI`).

Notes in

You can create single events to trigger from individual keys, or have layers and velocity fades by adding extra logic.

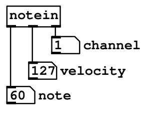


fig 3.21: MIDI note in

The `notein` object produces note number, velocity and channel values on its left, middle and right outlets. You may assign an object to listen to only one channel by giving it an argument from 1 to 15. Remember that note-off messages are equivalent to a note-on with zero velocity in many MIDI implementations and Pd follows this method. You therefore need to add extra logic before connecting an oscillator or sample player to `notein` so that zero valued MIDI notes are not played.

Notes out

Another object `noteout` sends MIDI to external devices. The first, second and third inlets set note number, velocity and channel respectively. The channel is 1 by default. Make sure you have something connected that can play back MIDI and set the patch shown in Fig. 3.22 running with its toggle switch. Every 200ms it produces a `C` on a random octave with a random velocity value between 0 and 127. Without further ado these could be sent to `noteout`, but it would cause each MIDI note to "hang", since we never send a note-off message. To properly construct MIDI notes you need `makenote`

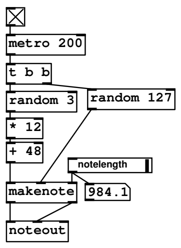


fig 3.22: MIDI note generation

which takes a note-number and velocity, and a duration (in milliseconds) as its third argument. After the duration has expired it automatically adds a note-off. If more than one physical MIDI port is enabled then `noteout` sends channels 1 to 16 to port 1 and channels 17 to 32 to port 2 etc.

Continuous controllers

Two MIDI input/output objects are provided to receive and send continuous controllers, `ctlin` and `ctlout`. Their three connections provide, or let you set, the controller value, controller number and MIDI channel. They can be instantiated with arguments, so `ctlin 10 1` picks up controller 10 (pan position) on MIDI channel 1.

MIDI to Frequency

Two numerical conversion utilities are provided to convert between MIDI note numbers and Hz. To get from MIDI to Hz use `mtof`. To convert a frequency in Hz to a MIDI note number use `ftom`.

Other MIDI objects

For pitchbend, program changes, system exclusive, aftertouch and other MIDI functions you may use any of the objects summarised in Tbl. 3.23. System exclusive messages may be sent by hand crafting raw MIDI bytes and outputting via the `midiout` object. Most follow the inlet and outlet template of `notein` and `noteout` having a channel as the last argument, except for `midiin` and `sysexin` which receive omni (all channels) data.

MIDI in object		MIDI out object	
Object	Function	Object	Function
`notein`	Get note data	`noteout`	Send note data.
`bendin`	Get pitchbend data −63 to +64	`bendout`	Send pitchbend data −64 to +64.
`pgmin`	Get program changes.	`pgmout`	Send program changes.
`ctlin`	Get continuous controller messages.	`ctlout`	Send continuous controller messages.
`touchin`	Get channel aftertouch data.	`touchout`	Send channel aftertouch data.
`polytouchin`	Polyphonic touch data in	`polytouchout`	Polyphonic touch output
`polytouchin`	Send polyphonic aftertouch.	`polytouchin`	Get polyphonic aftertouch.
`midiin`	Get unformatted raw MIDI	`midiout`	Send raw MIDI to device.
`sysexin`	Get system exclusive data	No output counterpart	Use `midiout` object

fig 3.23: List of MIDI objects

SECTION 3.6

Working with numbers

Arithmetic objects

Objects that operate on ordinary numbers to provide basic maths functions are summarised in Tbl. 3.24 All have hot left and cold right inlets and all take one argument that initialises the value otherwise received on the right inlet. Note the difference between arithmetic division with `/` and the `div` object. The modulo operator gives the remainder of dividing the left number by the right.

Object	Function
`+`	Add two floating point numbers
`-`	Subtract number on right inlet from number on left inlet
`/`	Divide lefthand number by number on right inlet
`*`	Multiply two floating point numbers
`div`	Integer divide, how many times the number on the right inlet divides exactly into the number on the left inlet
`mod`	Modulo, the smallest remainder of dividing the left number into any integer multiple of the right number

fig 3.24: Table of message arithmetic operators

Trigonometric maths objects

A summary of higher maths functions is given in Tbl. 3.25.

Random numbers

A useful ability is to make random numbers. The `random` object gives integers over the range given by its argument including zero, so `random 10` gives 10 possible values from 0 to 9.

Arithmetic example

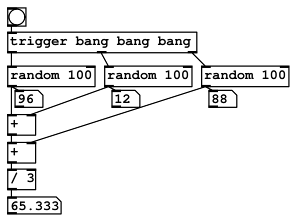


fig 3.26: Mean of three random floats

An example is given in Fig. 3.26 to show correct ordering in a patch to calculate the mean of three random numbers. We don't have to make every inlet hot, just ensure that everything arrives in the correct sequence by triggering the `random` objects properly. The first `random` (on the right) supplies the cold inlet of the lower `+`, the middle one to the cold inlet of the upper `+`. When the final (left) `random` is generated it passes to the hot inlet of the first `+`, which computes the sum and passes it to the second `+` hot inlet. Finally we divide by 3 to get the mean value.

Object	Function
cos	The cosine of a number given in radians. Domain: $-\pi/2$ to $+\pi/2$. Range: -1.0 to $+1.0$.
sin	The sine of a number in radians, domain $-\pi/2$ to $+\pi/2$, range -1.0 to $+1.0$
tan	Tangent of number given in radians. Range: 0.0 to ∞ at $\pm\pi/2$
atan	Arctangent of any number in domain $\pm\infty$ Range: $\pm\pi/2$
atan2	Arctangent of the quotient of two numbers in Cartesian plane. Domain: any floats representing X, Y pair. Range: angle in radians $\pm\pi$
exp	Exponential function e^x for any number. Range 0.0 to ∞
log	Natural log (base e) of any number. Domain: 0.0 to ∞. Range: $\pm\infty$ ($-\infty$ is -1000.0)
abs	Absolute value of any number. Domain $\pm\infty$. Range 0.0 to ∞
sqrt	The square root of any positive number. Domain 0.0 to ∞
pow	Exponentiate the left inlet to the power of the right inlet. Domain: positive left values only.

fig 3.25: Table of message trigonometric and higher math operators

Comparative objects

In Tbl. 3.27 you can see a summary of comparative objects. Output is either 1 or 0 depending on whether the comparison is true or false. All have hot left inlets and cold right inlets and can take an argument to initialise the righthand value.

Object	Function
>	True if the number at the left inlet is greater than the right inlet.
<	True if the number at the left inlet is less than the right inlet.
>=	True if the number at the left inlet is greater than or equal to the right inlet.
<=	True if the number at the left inlet is less than or equal to the right inlet.
==	True if the number at the left inlet is equal to the right inlet.
!=	True if the number at the left inlet is not equal to the right inlet

fig 3.27: List of comparative operators

Boolean logical objects

There are a whole bunch of logical objects in Pd including bitwise operations that work exactly like C code. Most of them aren't of much interest to us in this book, but we will mention the two important ones || and &&. The output of ||, logical OR, is true if either of its inputs are true. The output of &&, logical AND, is true only when both its inputs are true. In Pd any non-zero number is "true", so the logical inverter or "not" function is unnecessary because there are many ways of achieving this using other objects. For example, you can make a logical inverter by using != with 1 as its argument.

SECTION 3.7

Common idioms

There are design patterns that crop up frequently in all types of programming. Later we will look at abstraction and how to encapsulate code into new objects so you don't find yourself writing the same thing again and again. Here I will introduce a few very common patterns.

Constrained counting

We have already seen how to make a counter by repeatedly incrementing the value stored in a float box. To turn an increasing or decreasing counter into a cycle for repeated sequences there is an easier way than resetting the counter when it matches an upper limit, we wrap the numbers using mod. By inserting mod into the feedback path before the increment we can ensure the counter stays bounded. Further mod units can be added to the number stream to generate polyrhythmic sequences. You will frequently see variations on the idiom shown in Fig. 3.28. This is the way we produce multi-rate timebases for musical sequencers, rolling objects or machine sounds that have complex repetitive patterns.

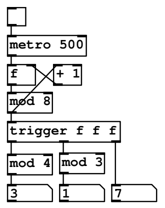


fig 3.28: Constrained counter.

Accumulator

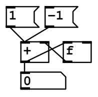


fig 3.29: Accumulator.

A similar construct to a counter is the accumulator or integrator. This reverses the positions of f and + to create an integrator that stores the sum of all previous number messages sent to it. Such an arrangement is useful for turning "up and down" messages from an input controller into a position. Whether to use a counter or accumulator is a subtle choice. Although you can change the increment step of the counter by placing a new value on the right inlet of + it will not take effect until the previous value in f has been used. An accumulator on the other hand can be made to jump different intervals immediately by the value sent to it. Note the important difference, an accumulator takes floats as an input while a counter takes bang messages.

Rounding

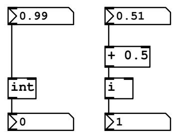


fig 3.30: Rounding

An integer function, int, also abbreviated i gives the whole part of a floating point number. This is a *truncation*, which just throws away any decimal digits. For positive numbers it gives the *floor* function, written $\lfloor x \rfloor$ which is the integer *less than or equal to* the input value. But take note of what happens for *negative* values, applying int to -3.4 will give 3.0, an integer *greater than or equal to* the input. Truncation is shown on the left of Fig. 3.30. To get a regular rounding for positive numbers, to pick the *closest* integer, use the method shown on the right side of Fig. 3.30. This will return 1 for an input of 0.5 or more and 0 for an input of 0.49999999 or less.

Scaling

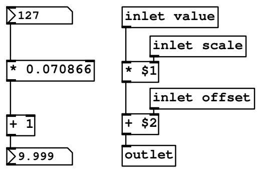


fig 3.31: Scaling

This is such a common idiom you will see it almost everywhere. Given a range of values such as 0 to 127 we may wish to map this onto another set of values, the domain, such as 1 to 10. This is the same as changing the slope and zero intersect of a line following $y = mx + c$. To work out the values you first obtain the bottom value or *offset*, in this case +1. Then a *multiplier* value is needed to scale for the upper value, which given an input of 127 would satisfy $10 = 1 + 127x$, so moving the offset we get $9 = 127x$, and dividing by 127 we get $x = 9/127$ or $x = 0.070866$. You can make a subpatch or an abstraction for this as shown in Fig. 6.1, but since only two objects are used it's more sensible to do scaling and offset as you need it.

Looping with until

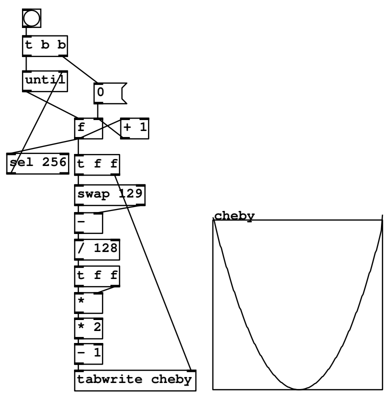


fig 3.32: Using until

Unfortunately, because it must be designed this way, until has the potential to cause a complete system lock-up. Be very careful to understand what you are doing with this. A bang message on the left inlet of until will set it producing bang messages as fast as the system can handle! These do not stop *until* a bang message is received on the right inlet. Its purpose is to behave as a fast loop construct performing message domain computation quickly. This way you can fill an entire wavetable or calculate a complex formula in the time it takes to process a single audio block. Always make sure the right inlet is connected to a valid terminating condition. In Fig. 3.32 you can see an example that computes the second Chebyshev polynomial according to $y = 2x^2 - 1$ for the range -1.0 to $+1.0$ and fills a 256 step table with the result. As soon as the bang button is pressed a counter is reset to zero and then until begins sending out bangs. These cause the counter to rapidly increment until select matches 256 whereupon a bang is sent to the right inlet of until stopping the process. All this will happen in a fraction of a millisecond. Meanwhile we use the counter output to calculate a Chebyshev curve and put it into the table.

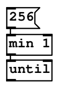


fig 3.33: for 256

A safer way to use until is shown in Fig. 3.33. If you know in advance that you want to perform a fixed number of operations then use it like a **for loop**. In this case you pass a non-zero float to the left inlet. There is no terminating condition, it stops when the specified number of bangs has been sent, 256 bangs in the example shown.

Message complement and inverse

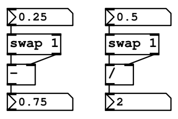


fig 3.34: Message reciprocal and inverse

Here is how we obtain the number that is $1 - x$ for any x. The *complement* of x is useful when you want to balance two numbers so they add up to a constant value, such as in panning. The swap object exchanges its inlet values, or any left inlet value with its first argument. Therefore, what happens with the left example of Fig. 3.34 is the - calculates $1 - x$, which for an input of 0.25 gives 0.75. Similarly the inverse of a float message $1/x$ can be calculated by replacing the - with a /.

Random selection

To choose one of several events at random a combination of random and select will generate a bang message on the select outlet corresponding to one of its arguments. With an initial argument of 4 random produces a *range* of 4 random integer numbers starting at 0, so we use select 0 1 2 3 to select amongst them. Each has an equal probability, so every outlet will be triggered 25% of the time on average.

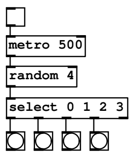


fig 3.35: Random select.

Weighted random selection

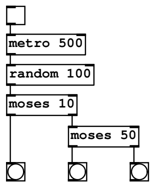


fig 3.36: Weighted random select.

A simple way to get a bunch of events with a certain probability distribution is to generate uniformly distributed numbers and stream them with moses. For example moses 10 sends integers greater than 9.0 to its right outlet. A cascade of moses objects will distribute them in a ratio over the combined outlets when the sum of all ratios equals the range of random numbers. The outlets of moses 10 distribute the numbers in the ratio 1 : 9. When the right outlet is further split by moses 50 as in Fig. 3.36 numbers in the range 0.0 to 100.0 are split in the ratio 10 : 40 : 50, and since the distribution of input numbers is uniform they are sent to one of three outlets with 10%, 40% and 50% probability.

Delay cascade

Sometimes we want a quick succession of bangs in a certain fixed timing pattern. An easy way to do this is to cascade `delay` objects. Each `delay 100` in Fig. 3.37 adds a delay of 100 milliseconds. Notice the abbrieved form of the object name is used.

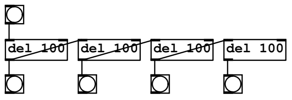


fig 3.37: Delay cascade.

Last float and averages

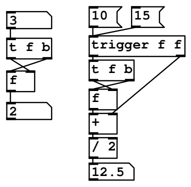


fig 3.38: Last value and averaging

If you have a stream of float values and want to keep the previous value to compare to the current one then the idiom shown on the left of Fig. 3.38 will do the job. Notice how a trigger is employed to first bang the *last* value stored in the float box and then update it with the current value via the right inlet. This can be turned into a simple "lowpass" or averaging filter for float messages as shown on the right of Fig. 3.38. If you add the previous value to the current one and divide by two you obtain the average. In the example shown the values were 10 followed by 15, resulting in $(10 + 15)/2 = 12.5$.

Running maximum (or minimum)

Giving `max` a very small argument and connecting whatever passes through it back to its right inlet gives us a way to keep track of the largest value. In Fig. 3.39 the greatest past value in the stream has been 35. Giving a very large argument to `min` provides the opposite behaviour for tracking a lowest value. If you need to reset the maximum or minimum tracker just send a very large or small float value to the cold inlet to start again.

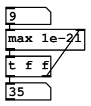


fig 3.39: Biggest so far

Float lowpass

Using only `*` and `+` as shown in Fig. 3.40 we can low pass filter a stream of float values. This is useful to smooth data from an external controller where values are occasionally anomalous. It follows the filter equation $y_n = Ax_n + Bx_{n-1}$. The strength of the filter is set by the ratio $A : B$. Both A and B should be between 0.0 and 1.0 and add up to 1.0. Note that this method will not converge on the exact input value, so you might like to follow it with `int` if you need numbers rounded to integer values.

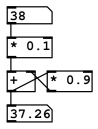


fig 3.40: Low pass for floats

CHAPTER 4
Pure Data Audio

SECTION 4.1

Audio objects

We have looked at Pd in enough detail now to move on to the next level. You have a basic grasp of dataflow programming and know how to make patches that process numbers and symbols. But why has no mention been made of audio yet? Surely it is the main purpose of our study? The reason for this is that audio signal processing is a little more complex in Pd than the numbers and symbols we have so far considered, so I wanted to leave this until now.

Audio connections

I already mentioned that there two kinds of objects and data for messages and signals. Corresponding to these there are two kinds of connections, audio connections and message connections. There is no need to do anything special to make the right kind of connection. When you connect two objects together Pd will work out what type of outlet you are attempting to connect to what kind of inlet and create the appropriate connection. If you try to connect an audio signal to a message inlet, then Pd will not let you, or it will complain if there is allowable but ambiguous connection. Audio objects always have a name ending with a tilde (∼) and the connections between them look fatter than ordinary message connections.

Blocks

The signal data travelling down audio cords is made of *samples*, single floating point values in a sequence that forms an audio signal. Samples are grouped together in *blocks.*

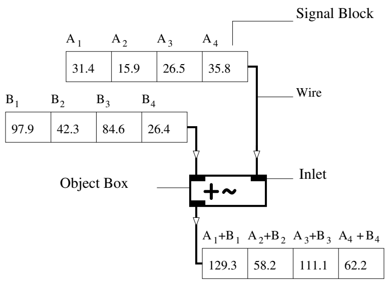


fig 4.1: Object processing data.

A block, sometimes also called a *vector*, typically has 64 samples inside it, but you can change this in certain circumstances. Objects operating on signal blocks behave like ordinary message objects, they can add, subtract, delay or store blocks of data, but do so by processing one whole block at a time. In Fig. 4.1 streams of blocks are fed to the two inlets. Blocks appearing at the outlet have values which are the sum of the corresponding values in the two input blocks. Because they process signals made of blocks, audio objects do a lot more work than objects that process messages.

Audio object CPU use

All the message objects we looked at in the last chapters only use CPU when event driven dataflow occurs, so most of the time they sit idle and consume no resources. Many of the boxes we put on our sound design canvases will be audio objects, so it's worth noting that they use up some CPU power just being idle. Whenever `compute audio` is switched on they are processing a constant stream of signal blocks, even if the blocks only contain zeros. Unlike messages which are processed in logical time, signals are processed synchronously with the soundcard sample rate. This *real-time* constraint means glitches will occur unless every signal object in the patch can be computed before the next block is sent out. Pd will not simply give up when this happens, it will struggle along trying to maintain real-time processing, so you need to listen carefully, as you hit the CPU limit of the computer you may hear crackles or pops. It is also worth knowing how audio computation relates to messages computation. Messages operations are executed at the beginning of each pass of audio block processing, so a patch where audio depends on message operations which don't complete in time will also fail to produce correct output.

SECTION 4.2

Audio objects and principles

There are a few ways that audio objects differ from message objects so let's look at those rules now before starting to create sounds.

Fanout and merging

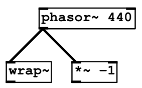


fig 4.2: Signal fanout is Okay.

You can connect the same signal outlet to as many other audio signal inlets as you like, and blocks are sent in an order which corresponds to the creation of the connections, much like message connections. But unlike messages, most of the time this will have no effect whatsoever, so you can treat audio signals that fan out as if they were perfect simultaneous copies. Very seldom you may meet rare and interesting problems, especially with delays and feedback, that can be fixed by reordering audio signals (see Chapter 7 of Puckette, "Theory and technique" regarding time shifts and block delays).

When several signal connections all come into the same signal inlet that's also fine. In this case they are implicitly summed, so you may need to scale your signal to reduce its range again at the output of the object. You can connect as many signals to the same inlet as you like, but sometimes it makes a patch easier to understand if you explicitly sum them with a +~ unit.

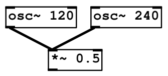


fig 4.3: Merging signals is Okay.

Time and resolution

Time is measured in seconds, milliseconds (one thousandth of a second, written 1ms) or samples. Most Pd times are in ms. Where time is measured in samples this depends on the sampling rate of the program or the sound card of the computer system on which it runs. The current sample rate is returned by the `samplerate~` object. Typically a sample is 1/44100th of a second and is the smallest unit of time that can be measured as a signal. But the time resolution also depends on the object doing the computation. For example `metro` and `vline~` are able to deal in fractions of a millisecond, even less than one sample. Timing irregularities can occur where some objects are only accurate to one block boundary and some are not.

Audio signal block to messages

To see the contents of a signal block we can take a snapshot or an average. The `env~` object provides the RMS value of one block of audio data scaled 0 to 100 in dB, while `snapshot~` gives the instantaneous value of the last sample in the previous block. To view an entire block for debugging `print~` can be used. It accepts an audio signal and a bang message on the same inlet and prints the current audio block contents when banged.

Sending and receiving audio signals

Audio equivalents of `send` and `receive` are written `send~` and `receive~`, with shortened forms `s~` and `r~` . Unlike message sends only one audio send can exist with a given name. If you want to create a signal bus with many to one connectivity use `throw~` and `catch~` instead. Within subpatches and abstractions we use the signal objects `inlet~` and `outlet~` to create inlets and outlets.

Audio generators

Only a few objects are signal sources. The most important and simple one is the `phasor~`. This outputs an asymmetrical periodic ramp wave and is used at the heart of many other digital oscillators we are going to make. Its left inlet specifies the frequency in Hz, and its right inlet sets the phase, between 0.0 and 1.0. The first and only argument is for frequency, so a typical instance of a phasor looks like `phasor~ 110`. For sinusoidal waveforms we can use `osc~` . Again, frequency and phase are set by the left and right inlets, or frequency is set by the creation parameter. A sinusoidal oscillator at concert A pitch is defined by `osc~ 440`. White noise is another commonly used source in sound design. The noise generator in Pd is simply `noise~` and has no creation arguments. Its output is in the range -1.0 to 1.0. Looped waveforms stored in an array can be used to implement *wavetable* synthesis using the `tabosc4~` object. This is a 4 point interpolating table oscillator and requires an array that is a power of 2, plus 3 (eg. 0 to 258) in order to work properly. It can be instantiated like `phasor~` or `osc~` with a frequency argument. A table oscillator running at 3kHz is shown in Fig. 4.4. It takes the waveform stored in array `A` and loops around this at the frequency given by its argument or left inlet value. To make sound samplers we need to read and write audio data from an array. The index to `tabread~` and its interpolating friend `tabread4~` is a sample number, so you need to supply a signal with the correct slope and magnitude to get the proper playback rate. You can use the special `set` message to reassign `tabread4~` to read from another

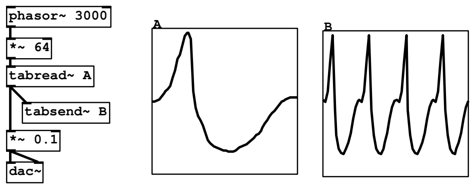


fig 4.4: Table oscillator

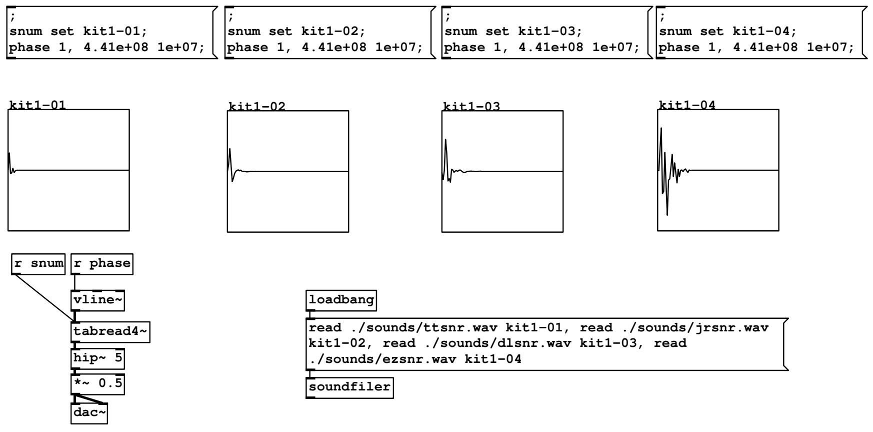


fig 4.5: Sample replay from arrays

array. The message boxes in Fig. 4.5 allow a single object to play back from more than one sample table. First the target array is given via a message to `snum`, and then a message is sent to `phase` which sets `vline~` moving up at 44100 samples per second. The arrays are initially loaded, using a multi-part message, from a `sounds` folder in the current patch directory.

Audio line objects

For signal rate control data the `line~` object is useful. It is generally programmed with a sequence of lists. Each list consists of a pair of numbers, the first being a level to move to and the second number is the time in milliseconds to take getting there. The range is usually between 1.0 and 0.0 when used as an audio control signal, but it can be any value such as when using `line~` to index a table. A more versatile line object is called `vline~`, which we will meet in much more detail later. Amongst its advantages are very accurate sub-millisecond timing and the ability to read multi-segment lists in one go and to delay stages of movement. Both these objects are essential for constructing envelope generators and other control signals.

Audio input and output

Audio IO is achieved with the adc~ and dac~ objects. By default these offer two inlets or outlets for stereo operation, but you can request as many additional sound channels as your sound system will handle by giving them numerical arguments.

Example: A simple MIDI monosynth

Using the objects we've just discussed let's create a little MIDI keyboard controlled music synthesiser as shown in Fig. 4.6. Numbers appearing at the left outlet of notein control the frequency of an oscillator. MIDI numbers are converted to a Hertz frequency by mtof. The MIDI standard, or rather general adherence to it, is a bit woolly by allowing note-off to also be a note-on with a velocity of zero. Pd follows this definition, so when a key is released it produces a note with a zero velocity. For this simple example we remove it with stripnote, which only passes note-on messages when their velocity is greater than zero. The velocity value, ranging between 1 and 127 is scaled to between 0 and 1 in order to provide a rudimentary amplitude control.

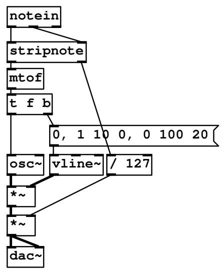


fig 4.6: MIDI note control

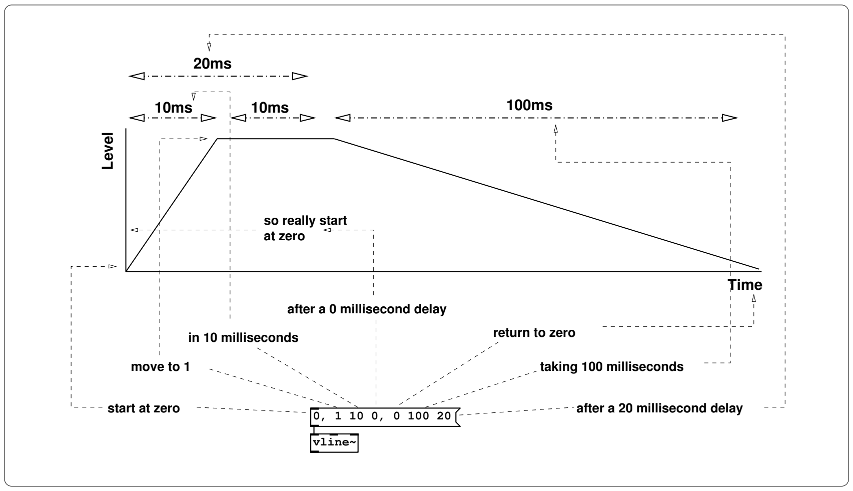


fig 4.7: Anatomy of vline message

So, here's a great place to elaborate on the anatomy of the message used to control vline~ as shown in Fig. 4.7. The syntax makes perfect sense, but sometimes it's hard to visualise without practice. The general form has three numbers per list. It says: *"go to some value"*, given by the first number, then *"take a certain time to get there"*, which is the second number in each list. The last number in the list is a time to wait before executing the command, so it adds an extra *wait for a time before doing it"*. What makes `vline~` cool is you can send a sequence of list messages in any order, and so long as they make temporal sense then `vline~` will execute them all. This means you can make very complex control envelopes. Any missing arguments in a list are dropped in right to left order, so a valid exception is seen in the first element of Fig. 4.7 where a single 0 means *"jump immediately to zero"* (don't bother to wait or take any time getting there).

Audio filter objects

Six or seven filters are used in this book. We will not look at them in much detail until we need to because there is a lot to say about their usage in each case. Simple one pole and one zero real filters are given by `rpole~` and `rzero~`. Complex one pole and one zero filters are `cpole~` and `czero~`. A static biquad filter `biquad~` also comes with a selection of helper objects to calculate coefficients for common configurations and `lop~`, `hip~`, and `bp~ 1` provide the standard low, high and bandpass responses. These are easy to use and allow message rate control of their cutoff frequencies and, in the case of bandpass, resonance. The first and only argument of the low and high pass filters is frequency, so typical instances may look like `lop~ 500` and `hip~ 500`. Bandpass takes a second parameter for resonance like this `bp~ 100 3`. Fast signal rate control of cutoff is possible using the versatile `vcf~` "voltage controlled filter". Its first argument is cutoff frequency and its second argument is resonance, so you might use it like `vcf~ 100 2`. With high resonances this provides a sharp filter that can give narrow bands. An even more colourful filter for use in music synthesiser designs is available as an external called `moog~`, which provides a classic design that can self oscillate.

Audio arithmetic objects

Audio signal objects for simple arithmetic are summarised in Tbl. 4.8.

Object	Function
`+~`	Add two signals (either input will also accept a message)
`-~`	Subtract righthand signal from lefthand signal
`/~`	Divide lefthand signal by right signal
`*~`	Signal multiplication
`wrap~`	Signal wrap, constrain any signal between 0.0 and 1.0

fig 4.8: List of arithmetic operators

Trigonometric and math objects

A summary of higher maths functions is given in Tbl. 4.9. Some signal units are abstractions defined in terms of more elementary intrinsic objects and those marked * are only available through external libraries in some Pd versions.

Object	Function
cos~	Signal version of cosine function. Domain: −1.0 `to` + 1.0. Note the input domain is "rotation normalised"
sin~	Not intrinsic but defined in terms of signal cosine by subtracting 0.25 from the input.
atan~ *	Signal version of arctangent with normalised range.
log~	Signal version of natural log.
abs~ *	Signal version of abs.
sqrt~	A square root for signals.
q8_sqrt~	A fast square root with less accuracy.
pow~	Signal version of power function.

fig 4.9: List of trig and higher math operators

Audio delay objects

Delaying an audio signal requires us to create a memory buffer using `delwrite~`. Two arguments must be supplied at creation time, a unique name for the memory buffer and a maximum size in milliseconds. For example, `delwrite~ mydelay 500` creates a named delay buffer "mydelay" of size 500ms. This object can now be used to write audio data to the delay buffer through its left inlet. Getting delayed signals back from a buffer needs `delread~`. The only argument needed is the name of a buffer to read from, so `delread~ mydelay` will listen to the contents of `mydelay`. The delay time is set by a second argument, or by the left inlet. It can range from zero to the maximum buffer size. Setting a delay time larger than the buffer results in a delay of the maximum size. It is not possible to alter the maximum size of a `delwrite~` buffer once created. But it is possible to change the delay time of `delread~` for chorus and other effects. This often results in clicks and pops [1] so we have a `vd~` variable-delay object. Instead of moving the read point `vd~` changes the rate it reads the buffer, so we get tape echo and Doppler shift type effects. Using `vd~` is as easy as before, create an object that reads from a named buffer like `vd~ mydelay`. The left inlet (or argument following the name) sets the delay time.

[1]Hearing clicks when moving a delay read point is normal, not a bug. There is no reason to assume that wavforms will align nicely once we jump to a new location in the buffer. An advanced solution crossfades between more than one buffer.

CHAPTER 5 Abstraction

λ

SECTION 5.1
Subpatches

Any patch canvas can contain *subpatches* which have their own canvas but reside within the same file as the main patch, called the *parent*. They have inlets and outlets, which you define, so they behave very much like regular objects. When you save a canvas all subpatches that belong to it are automatically saved. A subpatch is just a neat way to hide code, it does not automatically offer the benefit of local scope[1].

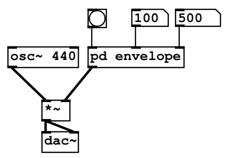


fig 5.1: Using an envelope subpatch

Any object that you create with a name beginning `pd` will be a subpatch. If we create a subpatch called `pd envelope` as seen in Fig. 5.1 a new canvas will appear and we can make `inlet` and `outlet` objects inside it as shown in Fig. 5.2. These appear as connections on the outside of the subpatch box in the same order they appear left to right inside the subpatch. I've given extra (optional) name parameters to the subpatch inlets and outlets. These are unnecessary, but when you have a subpatch with several inlets or outlets it's good to give them names to keep track of things and remind yourself of their function.

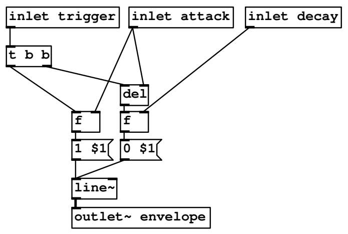


fig 5.2: Inside the envelope subpatch

To use `pd envelope` we supply a bang on the first inlet to trigger it, and two values for attack and decay. In Fig. 5.1 it modulates the output of an oscillator running at 440Hz before the signal is sent to `dac~`. The envelope has a trigger inlet for a message to bang two floats stored from the remaining inlets, one for the attack time in milliseconds and one for the decay time in milliseconds. The attack time also sets the period of a delay so that the decay portion of the envelope is not triggered until the attack part has finished. These values are substituted into the time parameter of a 2 element list for `line~`.

Copying subpatches

So long as we haven't used any objects requiring unique names any subpatch can be copied. Select `pd envelope` and hit `CTRL+D` to duplicate it. Having made

[1]As an advanced topic subpatches can be used as target name for dynamic patching commands or to hold datastructures.

one envelope generator it's a few simple steps to turn it into a MIDI mono synthesiser (shown in Fig. 5.3) based on an earlier example by replacing the `osc~` with a `phasor~` and adding a filter controlled by the second envelope in the range 0 to 2000Hz. Try duplicating the envelope again to add a pitch sweep to the synthesiser.

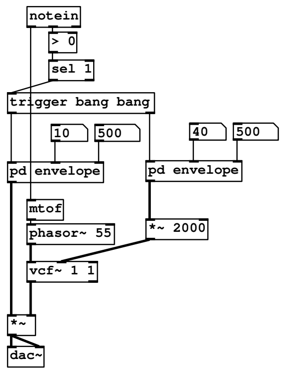


fig 5.3: Simple mono MIDI synth made using two copies of the same envelope subpatch

Deep subpatches

Consider an object giving us the vector magnitude of two numbers. This is the same as the hypotenuse c of a right angled triangle with opposite and adjacent sides a and b and has the formula $c = \sqrt{a^2 + b^2}$. There is no intrinsic object to compute this, so let's make our own subpatch to do the job as an exercise.

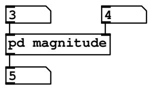


fig 5.4: Vector magnitude

We begin by creating a new object box and typing `pd magnitude` into it. A new blank canvas will immediately open for us to define the internals. Inside this new canvas create two new object boxes at the top by typing the word `inlet` into each. Create one more object box at the bottom as an `outlet`. Two input numbers a and b will come in through these inlets and the result c will go to the outlet.

When turning a formula into a dataflow patch it sometimes helps to think in reverse, from the bottom up towards the top. In words, c is the square root of the sum of two other terms, the square of a and the square of b. Begin by creating a `sqrt` object and connecting it to the outlet. Now create and connect a `+` object to the inlet of the `sqrt`. All we need to complete the example is an object that gives us the square of a number. We will define our own as a way to show that subpatches can contain other subpatches. And in fact this can go as deep as you like. It is one of the *principles of abstraction*, that we can define new

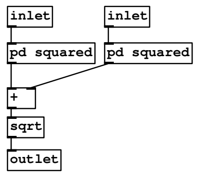


fig 5.5: Subpatch calculates $\sqrt{a^2 + b^2}$

objects, build bigger objects from those and still bigger objects in turn. Make a new object `pd squared` and when the canvas opens add the parts shown in Fig. 5.6.

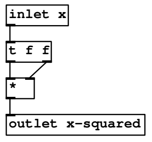


fig 5.6: Subpatch to compute x^2

To square a number you multiply it by itself. Remember why we use a trigger to split the input before sending it to each inlet of the multiply? We must respect evaluation order, so the trigger here distributes both copies of its input from right to left, the "cold" right inlet of `*` is filled first, then the "hot" left inlet. Close this canvas and connect up your new `pd squared` subpatch. Notice it now has an inlet and outlet on its box. Since we need two of them duplicate it by selecting then hitting `CTRL+D` on the keyboard. Your complete subpatch to calculate magnitude should look like Fig. 5.5. Close this canvas to return to the original topmost level and see `pd magnitude` now defined with two inlets and one outlet. Connect some number boxes to these as in Fig. 5.4 and test it out.

Abstractions

An abstraction is something that distances an idea from an object, it captures the essence and generalises it. It makes it useful in other contexts. Superficially an abstraction is a subpatch that exists in a separate file, but there is more to it. Subpatches add modularity and make patches easier to understand, which is one good reason to use them. However, while a subpatch seems like a separate object it is still part of a larger thing. *Abstractions* are reusable components written in plain Pd, but with two important properties. They can be loaded many times by many patches and although the same code defines all instances each instance has a separate internal namespace. They can also take creation arguments, so you can create multiple instances each with a different behaviour by typing different creation arguments in the object box. Basically, they behave like regular programming functions that can be called by many other parts of the program in different ways.

Scope and $0

Some objects like arrays and send objects must have a unique identifier, otherwise the interpreter cannot be sure which one we are referring to. In programming we have the idea of *scope* which is like a frame of reference. If I am talking to Simon in the same room as Kate I don't need to use Kate's surname every time I speak. Simon assumes, from context, that the Kate I am referring to is the most immediate one. We say that Kate has local scope. If we create an array within a patch and call it `array1` then that's fine so long as only one copy of it exists.

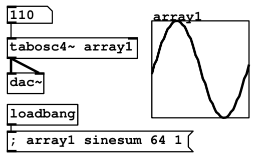


fig 5.7: Table oscillator patch

Consider the table oscillator patch in Fig. 5.7 which uses an array to hold a sine wave. There are three significant parts, a `tabosc4~` running at 110Hz, a table to hold one cycle of the waveform and an initialisation message to fill the table with a waveform. What if we want to make a multi-oscillator synthesiser using this method, but with a square wave in a one table and a triangle wave in another? We could make a subpatch of this arrangement and copy it, or just copy everything shown here within the main canvas. But if we do that without changing the array name, Pd will say;

```
warning: array1: multiply defined
warning: array1: multiply defined
```

The warning message is given twice because while checking the first array it notices another one with the same name, then later, while checking the duplicate array, it notices the first one has the same name. This is a serious warning and if we ignore it erratic, ill defined behaviour will result. We could rename each array we create as `array1`, `array2`, `array3` etc, but that becomes tedious. What we can to do is make the table oscillator an abstraction and give the array a special name that will give it local scope. To do this, select everything with `CTRL+E, CTRL+A` and make a new file from the file menu (Or you can use `CTRL+N` as a shortcut to make a new canvas). Paste the objects into the new canvas with `CTRL+V` and save it as `my-tabosc.pd` in a directory called `tableocillator`. The name of the directory isn't important, but it is important that we know where this abstraction lives so that other patches that will use it can find it. Now create another new blank file and save it as `wavetablesynth` in the *same* directory as the abstraction. This is a patch that will use the abstraction. By default a patch can find any abstraction that lives in the same directory as itself.

SECTION 5.2

Instantiation

Create a new object in the empty patch and type `my-tabosc` in the object box. Now you have an instance of the abstraction. Open it just as you would edit a normal subpatch and make the following changes as shown in Fig. 5.8;

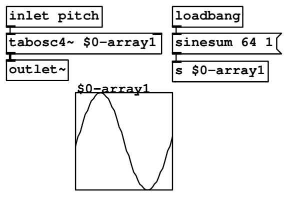


fig 5.8: Table oscillator abstraction

First we have replaced the number box with an inlet so that pitch data can come from outside the abstraction. Instead of a `dac~` the audio signal appears on an outlet we've provided. The most important change is the name of the array. Changing it to `$0-array1` gives it a special property. Adding the `$0-` prefix makes it local to the abstraction because at runtime, `$0-` is replaced by a unique per instance number. Of course we have renamed the array referenced by `tabosc4~` too. Notice another slight change in the table initialisation code, the message to create a sine wave is sent explicitly through a `send` because `$0-` inside a message box is treated in a different way.

SECTION 5.3
Editing

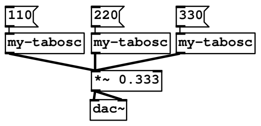


fig 5.9: Three harmonics using the table oscillator abstraction

Now that we have an abstracted table oscillator let's instantiate a few copies. In Fig. 5.9 there are three copies. Notice that no error messages appear at the console, as far as Pd is concerned each table is now unique. There is something important to note here though. If you open one of the abstraction instances and begin to edit it the changes you make will immediately take effect as with a subpatch, but they will only affect that instance. Not until you save an edited abstraction do the changes take place in *all* instances of the abstraction. Unlike subpatches, abstractions will not automatically be saved along with their parent patch and must be saved explicitly. Always be extra careful when editing abstractions to consider what the effects will be on all patches that use them. As you begin to build a library of reusable abstractions you may sometimes make a change for the benefit of one project that breaks another. How do you get around this problem? The answer is to develop a disciplined use of namespaces, prefixing each abstraction with something unique until you are sure you have a finished, general version that can used in all patches and will not change any more. It is also good practice to write help files for your abstractions. A file in the same directory as an abstraction, with the same name but ending `-help.pd` will be displayed when using the object help facility.

SECTION 5.4
Parameters

Making local data and variables is only one of the benefits of abstraction. A far more powerful property is that an abstraction passes any parameters given as creation arguments through local variables `$1, $2, $3`... In traditional programming terms this behaviour is more like a function than a code block. Each instance of an abstraction can be created with completely different initial arguments. Let's see this in action by modifying our table oscillator to take arguments for initial frequency and waveform. In Fig. 5.10 we see several interesting changes. Firstly, there are two `float` boxes that have `$n` parameters. You can use as many of these as you like and each of them will contain the nth creation parameter. They are all banged when the abstraction is loaded by the `loadbang`. The first sets the initial pitch of the oscillator, though of course this can still be over-ridden by later messages at the pitch inlet. The second activates one of three messages via `select` which contain harmonic series of square, sawtooth and sine waves respectively.

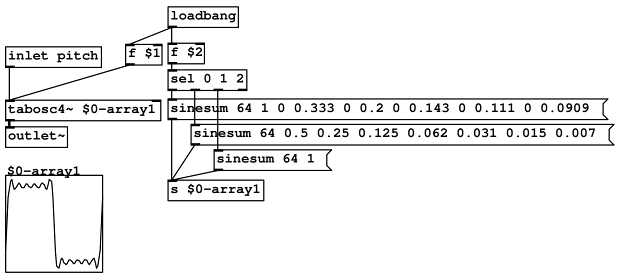

loadbang
inlet pitch
f $1
f $2
sel 0 1 2
tabosc4~ $0-array1
sinesum 64 1 0 0.333 0 0.2 0 0.143 0 0.111 0 0.0909
outlet~
sinesum 64 0.5 0.25 0.125 0.062 0.031 0.015 0.007
sinesum 64 1
$0-array1
s $0-array1

fig 5.10: Table oscillator abstraction with initialised frequency and shape.

SECTION 5.5

Defaults and states

A quick word about default parameters. Try creating some instances of the abstraction in Fig. 5.10 (shown as `my-tabsosc2` in Fig. 5.11)[2]. Give one a first parameter of 100Hz but no second parameter. What happens is useful, the missing parameter is taken to be zero. That's because float defaults to zero for an undefined argument. That's fine most of the time, because you can arrange for a zero to produce the behaviour you want. But, what happens if you create the object with no parameters at all? The frequency is set to 0Hz of course, which is probably useful behaviour, but let's say we wanted to have the oscillator start at 440Hz when the pitch is unspecified. You can do this with sel 0 so that zero value floats trigger a message with the desired default. Be careful choosing default behaviours for abstractions, they are one of the most common causes of problems later when the defaults that seemed good in one case are wrong in another. Another important point pertains to initial parameters of GUI components, which will be clearer in just a moment as we consider abstractions with built in interfaces. Any object that persistently maintains state (keeps its value between saves and loads) will be the same for *all* instances of the abstraction loaded. It can only have one set of values (those saved in the abstraction file). In other words it is the abstraction *class* that holds state, not the object instances. This is annoying when you have several instances of the same abstraction in a patch and want them to individually maintain persistent state. To do this you need a state saving wrapper like memento or sssad, but that is a bit beyond the scope of this textbook.

[2]The graphs with connections to them shown here, and elsewhere in the book, are abstractions that contain everything necsessary to display a small time or spectrum graph from signals received at an inlet. This is done to save space by not showing this in every diagram.

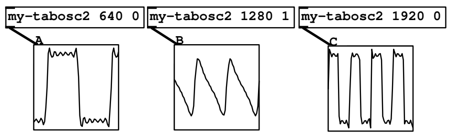


fig 5.11: Three different waveforms and frequencies from the same table oscillator abstraction

SECTION 5.6

Common abstraction techniques

Here are a few tricks regularly used with abstractions and subpatches. With these you can create neat and tidy patches and manage large projects made of reusable general components.

Graph On Parent

It's easy to build nice looking interfaces in Pd using GUI components like sliders and buttons. As a rule it is best to collect all interface components for an application together in one place and send the values to where they are needed deeper within subpatches. At some point it's necessary to expose the interface to the user, so that when an object is created it appears with a selection of GUI components laid out in a neat way.

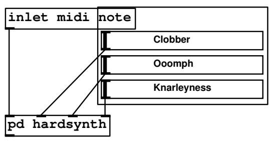


fig 5.12: Graph on parent synth

"Graph on Parent" (or GOP) is a property of the canvas which lets you see inside from outside the object box. Normal objects like oscillators are not visible, but GUI components, including graphs are. GOP abstractions can be nested, so that controls exposed in one abstraction are visible in a higher abstraction if it is also set to be GOP. In Fig. 5.12 we see a subpatch which is a MIDI synthesiser with three controls. We have added three sliders and connected them to the synth. Now we want to make this abstraction, called `GOP-hardsynth`, into a GOP abstraction that reveals the controls. Click anywhere on a blank part of the canvas, choose `properties` and activate the GOP toggle button. A frame will appear in the middle of the canvas. In the canvas properties box, set the size to $width = 140$ and $height = 80$, which will nicely frame three standard size sliders with a little border. Move the sliders into the frame, save the abstraction and exit.

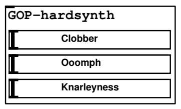


fig 5.13: Appearance of a GOP abstraction

Here is what the abstraction looks like when you create an instance (Fig. 5.13). Notice that the name of the abstraction appears at the top, which is why we left a little top margin to give this space. Although the inlet box partly enters the frame in Fig. 5.12 it cannot be seen in the abstraction instance because only GUI elements are displayed. Coloured *canvases*[3] also appear in GOP abstractions so if you want decorations they can be used to make things prettier. Any canvases appear above the name in the drawing order so if you want to hide the name make a canvas that fills up the whole GOP window. The abstraction name can be turned off altogether from the `properties` menu by activating `hide object name and arguments`.

Using list inputs

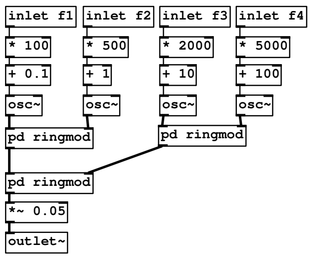


fig 5.14: Preconditioning normalised inlets

The patch in Fig. 5.14 is a fairly arbitrary example (a 4 source cross ring modulator). It's the kind of thing you might develop while working on a sound or composition. This is the way you might construct a patch during initial experiments, with a separate inlet for each parameter you want to modify. There are four inlets in this case, one for each different frequency that goes into the modulator stages. The first trick to take note of is the control pre-conditioners all lined up nicely at the top. These set the range and offset of each parameter so we can use uniform controls as explained below.

Packing and unpacking

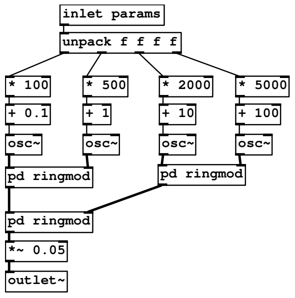
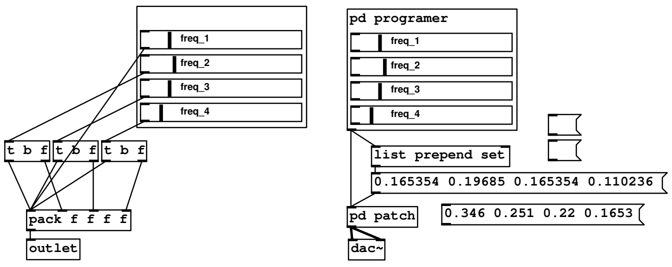


fig 5.15: Using a list input

What we've done here in Fig. 5.15 is simply replace the inlets with a single inlet that carries a list. The list is then unpacked into its individual members which are distributed to each internal parameter. Remember that lists are unpacked right to left, so if there was any computational order that needed taking care of you should start from the rightmost value and move left. This modification to the patch means we can use the flexible arrangement shown in Fig. 5.16 called a "programmer". It's just a collection of normalised sliders connected to a `pack` object so that a new list is transmitted each time a fader is moved. In order to do this it is necessary to insert `trigger bang float` objects between each slider as shown in Fig. 5.16 (left). These go on all but the far left inlet. Doing so ensures that the float value is loaded into `pack` before all the values are sent again. By prepending

[3]Here the word "canvas" is just used to mean a decorative background, different from the regular meaning of patch window.



(a) Packing a list (b) Making a programmer
fig 5.16: Packing and using parameter lists

the keyword `set` to a list, a message box that receives it will store those values. Now we have a way of creating patch presets, because the message box always contains a snapshot of the current fader values. You can see in Fig. 5.16 (right) some empty messages ready to be filled and one that's been copied ready to use later as a preset.

Control normalisation

Most patches require different parameter sets with some control ranges between 0.0 and 1.0, maybe some between 0.0 and 20000, maybe some bipolar ones −100.0 to +100.0 and so on. But all the sliders in the interface of Fig. 5.17 have ranges from 0.0 to 1.0. We say the control surface is *normalised*.

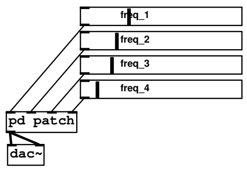


fig 5.17: All faders are normalised 0.0 to 1.0

If you build an interface where the input parameters have mixed ranges it can get confusing. It means you generally need a customised set of sliders for each patch. A better alternative is to normalise the controls, making each input range 0.0 to 1.0 and then adapting the control ranges as required inside the patch. Pre-conditioning means adapting the input parameters to best fit the synthesis pamaters. Normalisation is just one of the tasks carried out at this stage. Occasionally you will see a `log` or `sqrt` used to adjust the parameter curves. Pre-conditioning operations belong together as close to where the control signals are to be used as possible, They nearly always follow the same pattern, multiplier, then offset, then curve adjustment.

Summation chains

Sometimes when you have a lot of subpatches that will be summed to produce an output it's nicer to be able to stack them vertically instead of having many connections going to one place. Giving each an inlet (as in Fig. 5.18) and placing a `+~` object as part of the subpatch makes for easier to read patches.

Routed inputs

A powerful way to assign parameters to destinations while making them human readable is to use `route`. Look at Fig. 5.19 to see how you can construct arbitrary

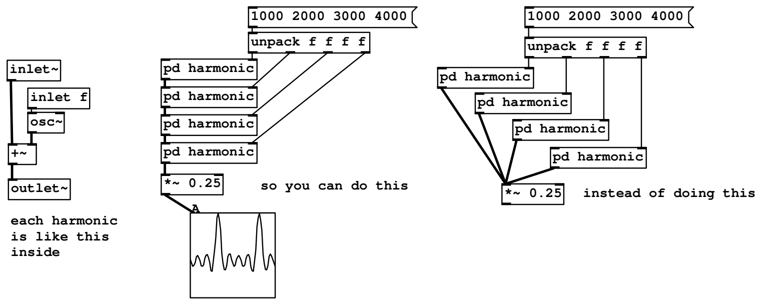


fig 5.18: Stacking subpatches that sum with an inlet

paths like URLs to break subpatches into individually addressable areas.

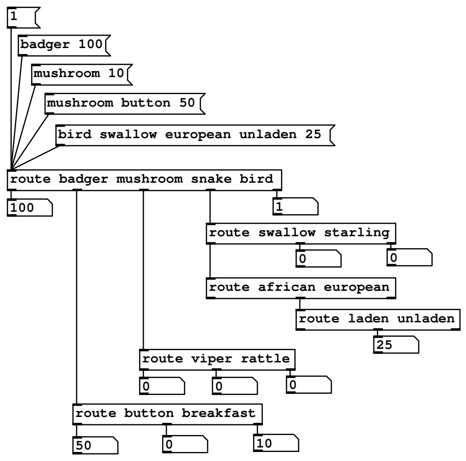


fig 5.19: Route can channel named parameters to a destination

CHAPTER 6
Shaping sound

The signal generators we've seen so far are the phasor, cosinusoidal oscillator, and noise source. While these alone seem limited they may be combined using shaping operations to produce a great many new signals. We are going to make transformations on waveforms, pushing them a little this way or that, moulding them into new things. This subject is dealt with in two sections, amplitude dependent shaping where the output depends only on current input values, and time dependent signal shaping where the output is a function of current and past signal values.

SECTION 6.1
Amplitude dependent signal shaping

Simple signal arithmetic

Arithmetic is at the foundation of signal processing. Examine many patches and you will find, on average, the most common object is the humble multiply, followed closely by addition. Just as all mathematics builds on a few simple arithmetic axioms complex DSP operations can be reduced to adds and multiplies. While it's rarely of practical use, it's worth noting that multiplication can be seen as repeated addition, so to multiply a signal by two we can connect it to both inlets of `+~` and it will be added to itself, which is the same as multiplying by two. The opposite of addition is subtraction. If you are subtracting a constant value from a signal it's okay to use `+~` but express the subtracted amount as a negative number, as with `+~ -0.5`, though of course there is a `-~` unit too. Addition and multiplication are commutative (symmetrical) operators, so it doesn't matter which way round you connect two signals to a unit. On the other hand, subtraction and division have ordered arguments, the right value is subtracted from, or is the divisor of, the left one. It is common to divide by a constant and so `*~` is generally used with an argument that's the reciprocal of the required divisor. For example, instead of dividing by two, multiply by half. There are two reasons for this. Firstly, divides were traditionally more expensive so many programmers are entrenched in the habit of avoiding divides where a multiply will do. Secondly, an accidental divide by zero traditionally causes problems, even crashing the program. Neither of these things are actually true of Pd running on a modern processor, but because of such legacies you'll find many algorithms written accordingly. Reserve divides for when you need to divide by a variable signal and multiply by decimal fractions everywhere else unless you need irrational numbers with high accuracy. This habit highlights the importance of the function and makes your patches easier to understand. Arithmetic operations are used to scale, shift and invert signals as the following examples illustrate.

A signal is scaled simply by multiplying it by a fixed amount, which changes the difference between the lowest and highest values and thus the peak to peak amplitude. This is seen in Fig. 6.1 where the signal from the oscillator is halved in amplitude.

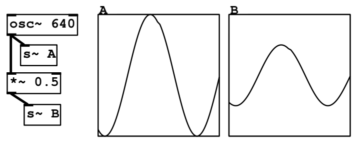


fig 6.1: Scaling a signal

Shifting involves moving a signal up or down in level by a constant. This affects the absolute amplitude in one direction only, so it is possible to distort a signal by pushing it outside the limits of the system, but it does not affect its peak to peak amplitude or apparent loudness since we cannot hear a constant (DC) offset. Shifting is normally used to place signals into the correct range for a subsequent operation, or, if the result of an operation yields a signal that isn't centered properly to correct it, so it swings about zero again. In Fig. 6.2 the cosine signal is shifted upwards by adding 0.5.

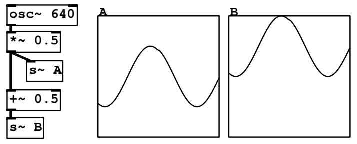


fig 6.2: Shifting a signal

In Fig. 6.3 a signal is inverted, reflecting it around the zero line, by multiplying by −1.0. It still crosses zero at the same places but its direction and magnitude is the opposite everywhere. Inverting a signal changes its phase by π, 180° or 0.5 in rotation normalised form, but that has no effect on how it sounds since we cannot hear absolute phase.

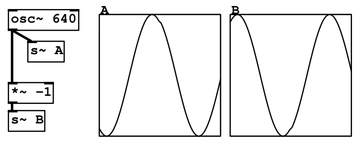


fig 6.3: Inverting a signal

The complement of a signal a in the range 0.0 to 1.0 is defined as $1 - a$. As the phasor in Fig. 6.4 moves upwards the complement moves downwards mirroring its movement. This is different from the inverse, it has the same direction as the inverse but retains the sign and is only defined for the positive range between 0.0 and 1.0. It is used frequently to obtain a control signal for amplitude or filter cutoff that moves in the opposite direction to another control signal.

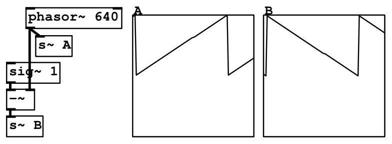


fig 6.4: Signal complement

For a signal a in the range 0.0 to x the reciprocal is defined as $1/a$. When a is very large then $1/a$ is close to zero, and when a is close to zero then $1/a$ is very large. Usually, since we are dealing with normalised signals, the largest input is $a = 1.0$, so because $1/1.0 = 1.0$ the reciprocal is also 1.0.

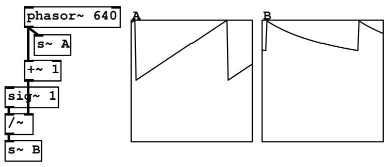


fig 6.5: Signal reciprocal

The graph of $1/a$ for a between 0.0 and 1.0 is a curve, so a typical use of the reciprocal is shown in Fig. 6.5. A curve is produced according to $1/(1 + a)$. Since the maximum amplitude of the divisor is 2.0 the minimum of the output signal is 0.5.

Limits

Sometimes we want to constrain a signal within a certain range. The `min~` unit outputs the minimum of its two inlets or arguments. Thus `min~ 1` is the minimum of one and whatever signal is on the left inlet, in other words it clamps the signal to a maximum value of one if it exceeds it. Conversely `max~ 0` returns the maximum of zero and its signal, which means that signals going below zero are clamped there forming a lower bound. You can see the effect of this on a cosine signal in Fig. 6.6.

Think about this carefully, the terminology seems to be reversed but it is correct. You use `max~` to create a minimum possible value and `min~` to create a maximum possible value. There is a slightly less confusing alternative `clip~` for situations where you don't want to adjust the limit using another signal. The left inlet of `clip~` is a signal and the remaining two inlets or arguments are the values of upper and lower limits, so for example `clip~ -0.5 0.5` will limit any signal into a range of one centered about zero.

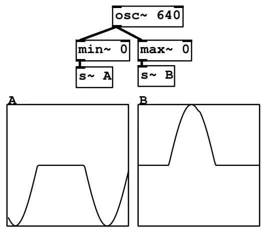


fig 6.6: Min and max of a signal

Wave shaping

Using these principles we can start with one waveform and apply operations to create others like square, triangle, pulse or any other shape. The choice of starting waveform is usually a phasor, since anything can be derived from it. Sometimes it's best to minimise the number of operations so a cosine wave is the best starting point.

One method of making a square wave is shown in Fig. 6.7. An ordinary cosine oscillator is multiplied by a large number and then clipped. If you picture a graph of a greatly magnified cosine waveform its slope will have become extremely steep, crossing through the area between -1.0 and 1.0 almost vertically. Once clipped to a normalised range what remains is

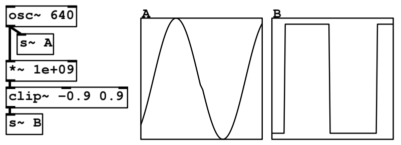


fig 6.7: Square wave

a square wave, limited to between 1.0 and −1.0 and crossing suddenly halfway through. This method produces a waveform that isn't band-limited, so when used in synthesis you should keep it to a fairly low frequency range to avoid aliasing.

A triangle wave moves up in a linear fashion just like a phasor, but when it reaches the peak it changes direction and returns to its lowest value at the same rate instead of jumping instantly back to zero. It is a little more complicated to understand than the square wave. We can make a signal travel more or less in a given time interval by multiplying it by a constant amount. If a signal is multiplied by 2.0 it will travel twice as far in the same time as it did before, so multiplication affects the slope of signals. Also, as we have just seen, multiplying a signal by −1.0 inverts it. That's another way of saying it reverses the slope, so the waveform now moves in the opposite direction. One way of making a triangle wave employs these two principles.

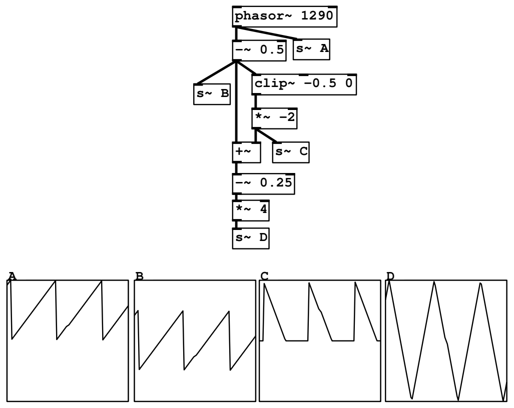


fig 6.8: Triangle

Starting with a phasor (graph *A*) at the top of Fig. 6.8, and shifting it down by 0.5 (graph *B*), the first half of it, from 0.0 to 0.5 is doing what we want. If we take half and isolate it with clip~ we can then multiply by −1.0 to change the slope, and by 2.0 to double the amplitude, which is the same as multiplying by −2.0. During the first half of the source phasor, between 0.5 and 1.0 the right branch produces a falling waveform (graph *C*). When we add that back to the other half, which is shifted down by 0.5 the sum is a triangle wave once normalised (graph *D*) .
An alternative formula for a triangle wave, which may be slightly easier to understand, uses min~ and is shown in Fig. 6.9. Starting with a phasor again, (graph *A*) and adding one to the inverse produces a negative moving phasor with the same sign but opposite phase (graph *B*). Taking the minima of these two signals gives us a triangle wave, positive with amplitude 0.5 (graph *C*). This is re-centered and normalised (graph *D*).

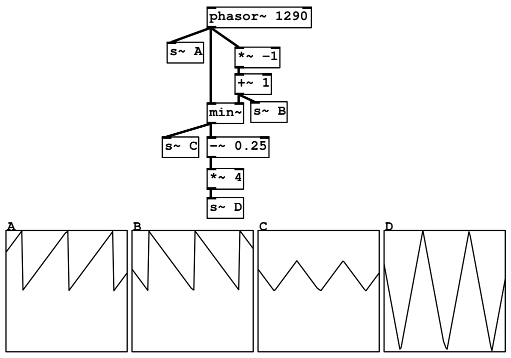


fig 6.9: Another way to make a triangle wave

Squaring and roots

One common function of a signal a is a^2, another way of writing $a \times a$. A multiplier is the easiest way to perform squaring. If you connect a signal to both inlets of a multiplier it is multiplied by itself. The effect of squaring a signal is twofold. It's amplitude is scaled as a function of it's own amplitude. Amplitude values that are already high are increased more, while values closer to zero are increased less. Another result is that the output signal is only positive. Since a minus times a minus gives a plus there are no squares that are negative. The reverse of this procedure is to determine a value r which if multiplied by itself gives the input a. We say r is the square root of a. Because finding square roots is a common DSP operation that requires a number of steps there's a built in sqrt~ object in Pd. Without creating complex (imaginary) numbers there are no square roots to negative numbers and so the output of sqrt~ is zero for these values. The effect of making the straight phasor line between 0.0 and 1.0 into

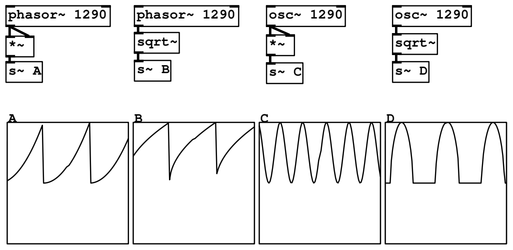


fig 6.10: Square roots

a curve is clear in Fig. 6.10 graph A, likewise the curve bends the other way for the square root in graph B. Remembering that a minus times a minus gives a plus you can see that whatever the sign of a signal appearing at both inlets of the multiplier, a positive signal is output in graph C. Making either sign of the cosine wave positive like this doubles the frequency. In graph D an absence of negative square roots produces a broken sequence of positive pulses, and the effect of the square root operation is to change the cosine curve to a parabolic (circular) curve (notice it is more rounded).

Curved envelopes

We frequently wish to create a curve from a rising or falling control signal in the range 0.0 to 1.0. Taking the square, third, fourth or higher powers produces increasingly steep curves, the class of *parabolic* curves. The quartic envelope is frequently used as a cheap approximation to natural decay curves. Similarly, taking successive square roots of a normalised signal will bend the curve the other way[1]. In Fig. 6.11 three identical line segments are generated each of

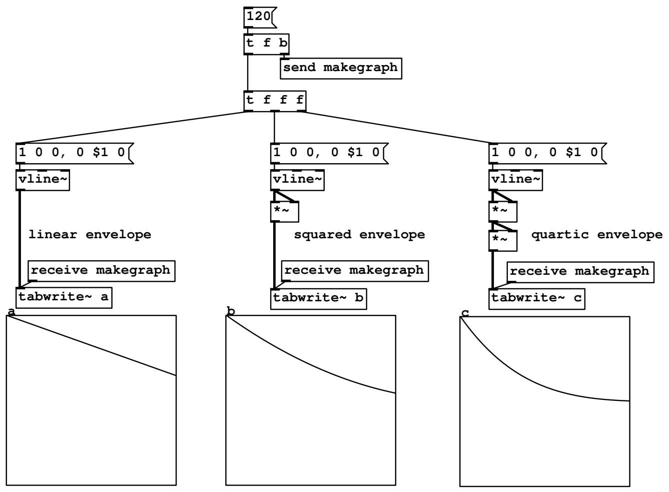


fig 6.11: Linear, squared and quartic decays

length 120ms. At the same time all `tabwrite~` objects are triggered so the graphs are synchronised. All curves take the same amount of time to reach zero, but as more squaring operations are added, raising the input to higher powers, the faster the curve decays during its initial stage.

SECTION 6.2

Periodic functions

A periodic function is bounded in range for an infinite domain. In other words, no matter how big the input value it comes back to the place it started from and repeats that range in a loop.

[1]See McCartney for other identities useful in making efficient natural envelopes.

Wrapping ranges

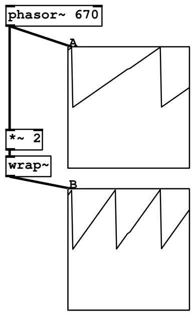


fig 6.12: Wrapping

The wrap~ object provides just such a behaviour. It is like a signal version of mod If the input a to wrap~ exceeds 1.0 then it returns $a - 1.0$. And if the input exceeds 2.0 it gives us $a - 2.0$. Wrap is the "fractional" part of a number in relation to a division, in this case the unit 1, $a - \lfloor a \rfloor$. Let's say we have a normalised phasor which is cycling up once per second. If we pass it through wrap~ it will be unaffected. A normalised phasor never exceeds 1.0 and so passes through unchanged. But if we double the amplitude of the phasor by multiplying by 2.0 and then wrap it something else happens as seen in Fig. 6.12.

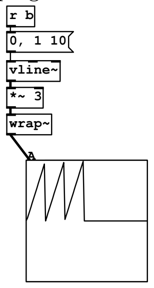


fig 6.13: Wrapping a line

Imagine the graph of a in a range of 0.0 to 2.0 drawn on tracing paper then the paper is cut into two strips of height 1.0 which are placed on top of one another. Each time the phasor passes 1.0 it is wrapped back to the bottom. Consequently the frequency doubles but it's peak amplitude stays at 1.0. This way we can create periodic functions from a steadily growing input, so a line that rises at a constant rate can be turned into a phasor with wrap~. Even more useful, we can obtain an exact number of phasor cycles in a certain time period by making the line rise at a particular rate. The vline~ in Fig. 6.13 moves from 0.0 to 1.0 in 10ms. Multiplying by 3 means it moves from 0.0 to 3.0 in 10ms, and wrapping it produces three phasor cycles in a period of $10/3 = 3.333$ms, giving a frequency of $1/3.333 \times 1000 = 300$Hz.

Cosine function

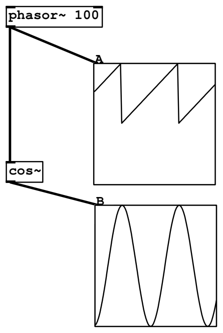


fig 6.14: Cosine of a phasor

The reason for saying that the phasor is the most primitive waveform is that even a cosinusoidal oscillator can be derived from it. Notice in Fig. 6.14 that although the phasor is always positive in the range 0.0 to 1.0 (unipolar) the cos~ operation produces a *bipolar* waveform in the range -1.0 to 1.0. One complete period of the cosine corresponds to 2π, 360°, or in rotation normalised form 1.0. When the phasor is at 0.0 the cosine is 1.0. When the phasor is at 0.25 the cosine crosses zero moving downwards. It reaches the bottom of its cycle when the phasor is 0.5. So there are two zero crossing points, one when the phasor is 0.25 and another when it is 0.75. When the phasor is 1.0 the cosine has completed a full cycle and returned to its original position.

SECTION 6.3

Other functions

From time to time we will use other functions like exponentiation, raising to a variable power, or doing the opposite by taking the log of a value. In each case we will examine the use in context. A very useful technique is that arbitrary curve shapes can be formed from *polynomials*

Polynomials

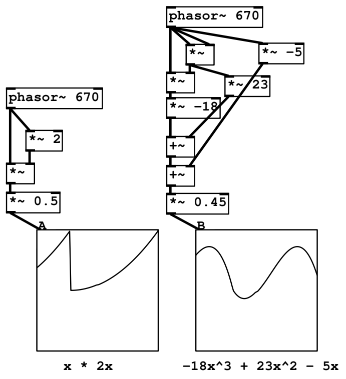


fig 6.15: Polynomials

A polynomial is expressed as a sum of different power terms. The graph of $2x^2$ gives a gently increasing slope and the graph of $18x^3+23x^2-5x$ shows a simple hump weighted towards the rear which could be useful for certain kinds of sound control envelope. There are some rules for making them. The number of times the curve can change direction is determined by which powers are summed. Each of these is called a term. A polynomial with some factor of the a^2 term can turn around once, so we say it has one *turning point*. Adding an a^3 term gives us two turning points and so on. The multiplier of each term is called the coefficient and sets the amount that term effects the shape. Polynomials are tricky to work with because it's not easy to find the coefficients to get a desired curve. The usual method is to start with a polynomial with a known shape and carefully tweak the coefficients to get the new shape you want. We will encounter some later like cubic polynomials that can be used to make natural sounding envelope curves.

Expressions

Expressions are objects with which you can write a single line of arbitrary processing code in a programmatic way. Each of many possible signal inlets x, y, z correspond to variables $\$V(x, y, z)$ in the expression and the result is returned at the outlet. This example shows how we generate a mix of two sine waves, one 5 times the frequency of the other. The available functions are very like those found in C and follow the maths syntax of most programming languages. Although expressions are very versatile they should only be used as a last resort when you cannot build from more primitive objects. They are less efficient than inbuilt objects and more difficult to read. The expression shown in Fig. 6.16 implements $Asin(2\pi\omega) + Bsin(10\pi\omega)$ for a periodic phasor ω and two mix coefficients where $B = 1 - A$. The equivalent patch made from primitives is shown at the bottom of Fig. 6.16.

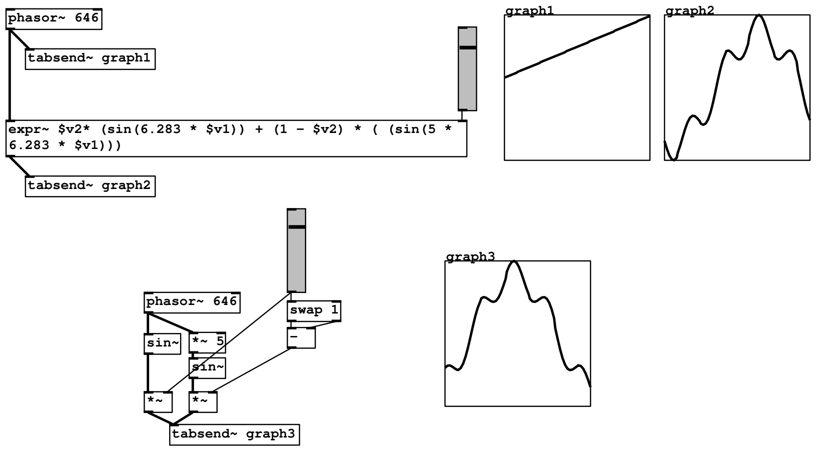


fig 6.16: Using an expression to create an audio signal function

SECTION 6.4
Time dependent signal shaping

So far we have considered ways to change the amplitude of a signal as a function of one or more other variables. These are all instantaneous changes which depend only on the current value of the input sample. If we want a signal to change its behaviour based on its previous features then we need to use time shaping.

Delay

To shift a signal in time we use a delay. Delays are at the heart of many important procedures like reverb, filters and chorusing. Unlike most other Pd operations, delays are used as two separate objects. The first is a write unit that works like send~ but sends the signal to an invisible area of memory. The second object is for reading from the same memory area after a certain time. So you always use delwrite~ and delread~ as pairs. The first argument to delwrite~ is a unique name for the delay and the second is the maximum memory (as time in milliseconds) to allocate. On it's own a delay just produces a perfect copy of an input signal a fixed number of milliseconds later. Here we see a 0.5ms pulse created by taking the square of a fast line from one to zero. The second graph shows the same waveform as the first but it happens 10ms later.

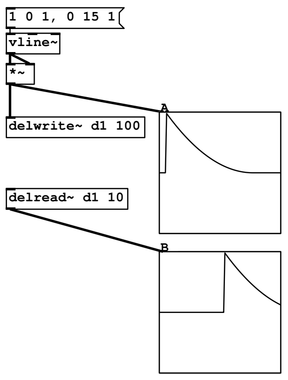


fig 6.17: Delay

Phase cancellation

Assuming that two adjacent cycles of a periodic waveform are largely the same then if we delay that periodic signal by time equal to half its period we have changed its phase by 180°. In the patch shown here the two signals are out of phase. Mixing the original signal back with a copy that is anti-phase annihilates both signals leaving nothing. In Fig. 6.18 a sinusoidal signal at 312Hz is sent to a delay **d1**. Since the input frequency is 312Hz its period is 3.2051ms, and half that is 1.60256ms. The delayed signal will be out of phase by half of the input signal period. What would happen if the delay were set so that the two signals were perfectly in phase? In that case instead of being zero the output would be a waveform with twice the input amplitude. For delay times between these two cases the output amplitude varies between 0.0 and 2.0. We can say for a given frequency component the output amplitude depends on the delay time. However, let's assume the delay is fixed and put it another way - for a given delay time the output amplitude depends on the input frequency. What we have created is a simple filter.

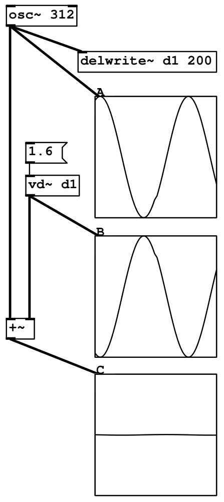


fig 6.18: Antiphase

Filters

When delay time and period coincide we call the loud part (twice the input amplitude) created by reinforcement a *pole*, and when the delay time equals half the period we call the quiet part where the waves cancel out a *zero*. Very basic but flexible filters are provided in Pd called rpole~ and rzero~. They are tricky to set up unless you learn a little more about DSP filter theory because the frequencies of the poles or zeros are determined by a normalised number that represents the range of 0Hz to $SR/2$Hz, where SR is the sampling rate of the patch. Simple filters can be understood by an equation governing how the output samples are computed as a function of the current or past samples. There are two kinds, those whose output depends only on past values of the *input*, which are called *finite impulse response* filters (FIR), and another type whose output depends on past input values **and** on past *output* values. In other words this kind has a feedback loop around the delay elements. Because the effect of a signal value could theoretically circulate forever we call this kind recursive or *infinite impulse response* filters (IIR).

User friendly filters

Filters may have many poles and zeros but instead of calculating these from delay times, sampling rates and wave periods we prefer to use filters designed with preset behaviours. The behaviour of a filter is determined by a built in calculator that works out the coefficients to set poles, zeros and feedback levels for one or more internal delays. Instead of poles and zeros we use a different terminology and talk about bands which are passed or stopped. A band has a center frequency, specified in Hz, the middle of the range where it has the most effect, and also a bandwidth which is the range of frequencies it operates over. Narrow bands affect fewer frequencies than wider bands. In many filter designs you can change the bandwidth and the frequency independently. Four commonly encountered filters are the low pass, high pass, band pass, and band cut or notch filter shown in Fig. 6.19 The graphs show the spectrum of white

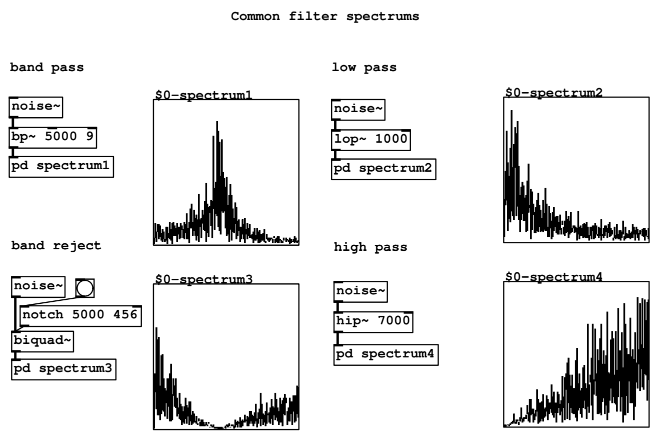


fig 6.19: Common user friendly filter shapes.

noise after it's been passed through each of the filters. The noise would normally fill up the graph evenly, so you can see how each of the filters cuts away at a different part of the spectrum. The high pass allows more signals above its centre frequency through than ones below. It is the opposite of the low pass which prefers low frequencies. The notch filter carves out a swathe of frequencies in the middle of the spectrum, which is the opposite of the band pass that allows a group of frequencies in the middle through but rejects those either side.

Integration

Another way of looking at the behaviour of filters is to consider their effect on the slope or phase of moving signals. One of the ways that recursive (IIR) filters can be used is like an accumulator. If the feedback is very high the current input is added to all previous ones. Integration is used to compute the area under a curve, so it can be useful for us to work out the total energy contained in a signal. It can also be used to shape waveforms.

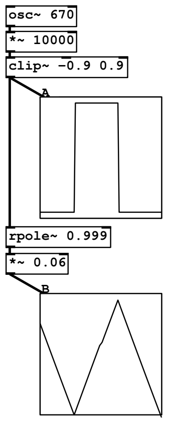


fig 6.20: Integration

Integrating a square wave gives us a triangle wave. If a constant signal value is given to an integrator its output will move up or down at a constant rate. In fact this is the basis of a phasor, so a filter can be seen as the most fundamental signal generator as well as a way to shape signals, thus we have come full circle and can see the words of the great master "It's all the same thing". A square wave is produced by the method shown in Fig. 6.7, first amplifying a cosinusoidal wave by a large value and then clipping it. As the square wave alternates between +1.0 and −1.0 the integrator output first slopes up at a constant rate, and then down at a constant rate. A scaling factor is added to place the resulting triangle wave within the bounds of the graph. Experiment with integrating a cosinusoidal wave. What happens? The integral of $\cos(x)$ is $\sin(x)$, or in other words we have shifted $\cos(x)$ by 90°. If the same operation is applied again, to a sine wave, we get back to a cosine wave out of phase with the first one, a shift of 180°. In other words the integral of $\sin(x)$ is $-\cos(x)$. This can be more properly written as a definite integral

$$\int \cos(x)\,dx = \sin(x) \tag{6.1}$$

or as

$$\int \sin(x)\,dx = -\cos(x) \tag{6.2}$$

Differentiation

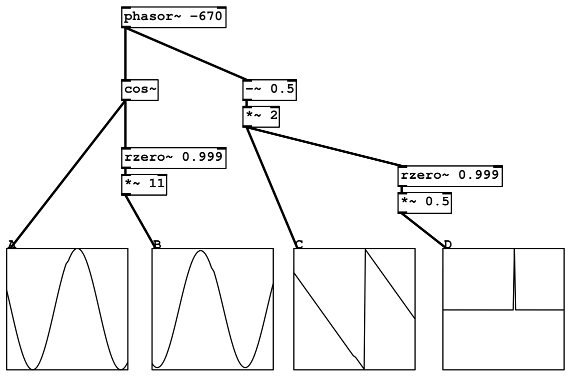


fig 6.21: Differentiation

The opposite of integrating a signal is differentiation. This gives us the instantaneous slope of a signal, or in other words the gradient of a line tangential to the signal. What do you suppose will be the effect of differentiating a cosine wave? The scaling factors in Fig. 6.21 are given for the benefit of the graphs. Perhaps you can see from the first graph that

$$\frac{d}{dx}\cos(x) = -\sin(x) \tag{6.3}$$

and

$$\frac{d}{dx}\sin(x) = \cos(x) \tag{6.4}$$

More useful perhaps, is the result of differentiating a phasor. While the phasor moves slowly its gradient is a small constant, but at the moment it suddenly returns the gradient is very high. So, differentiating a phasor is a way for us to obtain a brief impulse spike.

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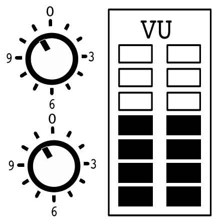
Puckette, M. (2007) “The theory and technique of electronic music” ISBN 978-981-270-077-3

Papers

McCartney, J. (1997) “Synthesis without Lookup Tables” Computer Music Journal 21(3)

CHAPTER 7
Pure Data essentials



This chapter will present some commonly used configurations for mixing, reading and writing files, communication and sequencing. You may want to build up a library of abstractions for things you do again and again, or to find existing ones from the pd-extended distribution. All the same, it helps to understand how these are built from primitive objects since you may wish to customise them to your own needs.

SECTION 7.1 Channel strip

For most work you will use Pd with multiple audio outlets and an external mixing desk. But you might find you want to develop software which implements its own mixing. All mixing desks consist of a few basic elements like gain controls, buses, panners and mute or channel select buttons. Here we introduce some basic concepts that can be plugged together to make complex mixers.

Signal switch

All we have to do to control the level of a signal is multiply it by a number between 0.0 and 1.0. The simplest form of this is a signal switch where we connect a toggle to one side of a `*~` and an audio signal to the other (Fig. 7.1). The toggle outputs either 1 or 0, so the signal is either on or off. You will use this frequently to temporarily block a signal. Because the toggle changes value abruptly it usually produces a click, so don't use this simple signal switch when recording audio, for that you must apply some smoothing as in the mute button below.

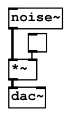


fig 7.1: Signal switch

Simple level control

To create a level fader start with a vertical slider and set its `properties` to a lower value of 0.0 and upper value of 1.0. In Fig. 7.2 the slider is connected to one inlet of `*~` and the signal to the other, just like the signal switch above except the slider gives a continuous change between 0.0 and 1.0. A number box displays the current fader value, 0.5 for a halfway position here. A sine oscillator at 40Hz provides a test signal. It is okay to mix messages and audio signals on opposite sides of `*~` like this, but because the slider generates messages any updates will only happen on each

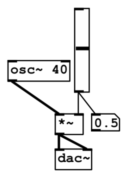


fig 7.2: Direct level control

block, normally every 64 samples. Move it up and down quickly and listen to the result. Fading is not perfectly smooth. You will hear a clicking sound when you move the slider. This *zipper noise* is caused by the level suddenly jumping to a new value on a block boundary.

Using a log law fader

The behaviour of slider objects can be changed. If you set its properties to log instead of linear then smaller values are spread out over a wider range and larger values are squashed into the upper part of the movement. This gives you a finer degree of control over level and is how most real mixing desks work. The smallest value the slider will output is 0.01. With its top value as 1.0 it will also output 1.0 when fully moved. Between these values it follows a logarithmic curve. When set to halfway it outputs a value of about 0.1 and at three quarters of full movement its output is a little over 0.3. It doesn't reach an output of 0.5 until nearly nine tenths of its full movement (shown in Fig. 7.3). This means half the output range is squashed into the final ten percent of the movement range, so be careful when you have this log law fader connected to a loud amplifier. Often log law faders are limited to constrain their range, which can be done with a `clip` unit.

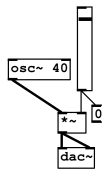


fig 7.3: log level control

MIDI fader

You won't always want to control a mix from Pd GUI sliders, sometimes you might wish to use a MIDI fader board or other external control surface. These generally provide a linear control signal in the range 0 to 127 in integer steps, which is also the default range of GUI sliders. To convert a MIDI controller message into the range 0.0 to 1.0 it is divided by 127 (the same as multiplying by 0.0078745) as shown in Fig. 7.4. The normalised output can be further scaled to a log curve, or multiplied by 100 to obtain a decibel scale and converted via the `dbtorms` object.

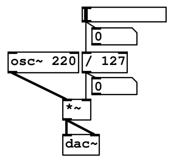


fig 7.4: Scaling a level

To connect the fader to an external MIDI device you need to add a `ctlin` object. The first outlet gives the current fader value, the second indicates the continuous controller number and the third provides the current MIDI channel. Volume messages are sent on controller number 7. We combine the outlets using `==` and `spigot` so that only volume control messages on a particular channel are passed to the fader. The patch shown in Fig. 7.5 has an audio inlet and outlet. It has an inlet to set the MIDI channel. It can be subpatched or abstracted to form one of several components in a complete MIDI controlled fader board.

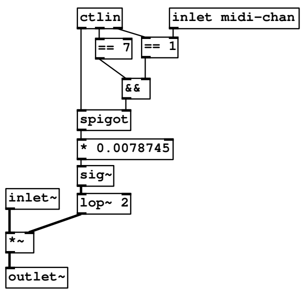


fig 7.5: MIDI level

Mute button and smooth fades

After carefully adjusting a level you may want to temporarily silence a channel without moving the slider. A *mute button* solves this problem. The fader value is stored at the cold inlet of a `*` while the left inlet receives a Boolean value from a toggle switch. The usual sense of a mute button is that the channel is silent when the mute is active, so first the toggle output is inverted. Some solutions to zipper noise use `line` or `line~` objects to interpolate the slider values. Using `line` is efficient but somewhat unsatisfactory since we are still interfacing a message to a signal and will hear clicks on each block boundary even though the jumps are smaller. Better is to use `line~`, but this can introduce corners into the control signal if the slider moves several times during a fade. A good way to obtain a smooth fade is to convert messages to a signal with `sig~` and then low pass filter it with `lop~`. A cutoff value of 1Hz will make a fade that smoothly adjusts over 1 second.

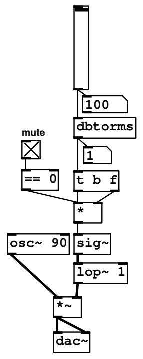


fig 7.6: mute switch

Panning

Pan is short for panorama, meaning a view in all directions. The purpose of a pan control is to place a *mono* or *point* source within a listening panorama. It should be distinguished from *balance* which positions a sound already containing stereo information. The field of an audio panorama is called the *image* and with plain old stereo we are limited to a theoretical *image width* of 180°. In practice a narrower width of 120° is used. Some software applications specify the pan position in degrees, but this is fairly meaningless unless you know precisely how the loudspeakers are arranged or whether the listener is using headphones. Mixing a stereo image for anything other than movie theatres is always a compromise to account for the unknown final listening arrangement. In movie sound however, the specifications of theatre PA systems are reliable enough to accurately predict the listeners experience.

Simple linear panner

In the simplest case a pan control provides for two speakers, left and right. It requires that an increase on one side has a corresponding decrease on the other. In the center position the sound is distributed equally to both loudspeakers. The pan patch in Fig. 7.7 shows a signal inlet and control message inlet at the top and two signal outlets at the bottom, one for the left channel and one for the right. Each outlet is preceded by a multiplier to set the level for that channel, so the patch is essentially two level controls in one. As with our level control, zipper noise is removed by converting control messages to a signal and then smoothing them with a filter. The resulting control signal, which is in the range 0.0 to

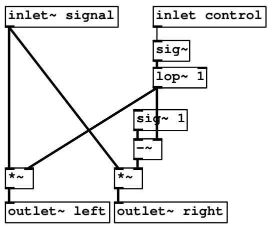


fig 7.7: simple panner

1.0, is fed to the left channel multiplier, while its complement (obtained by subtracting it from 1.0) governs the right side. With a control signal of 0.5 both sides are multiplied by 0.5. If the control signal moves to 0.75 then the opposing side will be 0.25. When the control signal reaches 1.0 the complement will be 0.0, so one side of the stereo image will be completely silent.

Square root panner

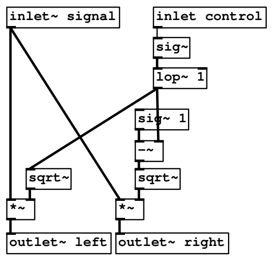


fig 7.8: root law panner

The problem with simple linear panning is that when a signal of amplitude 1.0 is divided in half and sent to two loudspeakers, so each receives an amplitude of 0.5, the result is quieter than sending an amplitude of 1.0 to only one speaker. This doesn't seem intuitive to begin with, but remember loudness is a consequence of sound power level, which is the square of amplitude. Let's say our amplitude of 1.0 represents a current of 10A. In one loudspeaker we get a power of $10^2 = 100$W. Now we send it to equally amongst two speakers, each receiving a current of 5A. The power from each speaker is therefore $5^2 = 25$W and the sum of them both is only 50W. The real loudness has halved! To remedy this we can modify the curve used to multiply each channel, giving it a new *taper*. Taking the square root of the control signal for one channel and the square root of the complement of the control signal for the other, gives panning that follows an *equal power law*. This has a 3dB amplitude increase in the center position.

Cosine panner

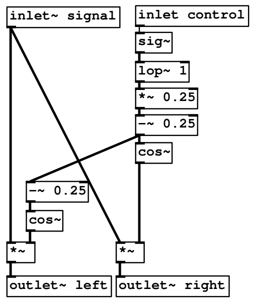


fig 7.9: cos-sin law panner

While the square root law panner gives a correct amplitude reduction for centre position it has a problem of its own. The curve of $\sqrt{A}$ is perpendicular to the x axis as it approaches it, so when adjusting the panning close to one side the image suddenly disappears completely from the other. An alternative taper follows the *sine-cosine law*. This also gives a smaller amplitude reduction in the centre position, but it approaches the edges of the image smoothly, at 45 degrees. The cosine panner is not only better in this regard but slightly cheaper in CPU cycles since it's easier to compute a cosine than a square root. It also mimics the placement of the source on a circle around the listener and is nice for classical music as an orchestra is generally arranged in a semicircle, however some engineers and producers prefer the root law panner because it has a nicer response around the center position and signals are rarely panned hard left or right.

Fig. 7.10 shows the taper of each panning law. You can see that the linear method is 3dB lower than the others in the centre position and that the root and cosine laws have different approaches at the edge of the image.

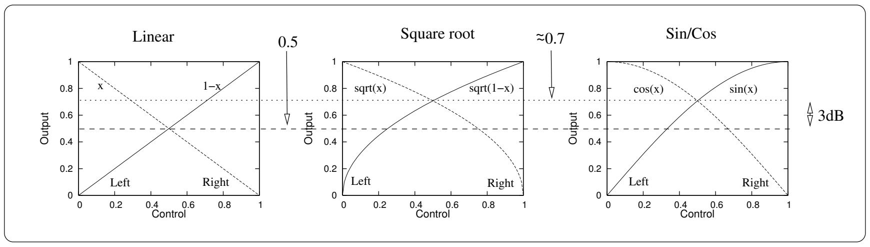


fig 7.10: Linear, root and sin/cos panning laws

Combining the cosine panner patch with a `ctlin` we now have a MIDI controlled pan unit to add to the MIDI controlled fader. Pan information is sent on controller number 10, with 64 representing the centre position. Once again an inlet is provided to select the MIDI channel the patch responds to. You may like to expand this idea into a complete MIDI fader board by adding a mute, bus outlet and auxiliary send/return loop. It might be a good solution to combine the level control, panning, mute and routing into a single abstraction that takes the desired MIDI channel and output bus as creation arguments. Remember to use dollar notation to create local variables if you intend to override MIDI control with duplicate controls from GUI objects.

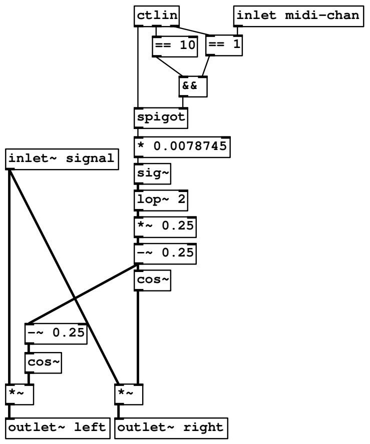


fig 7.11: MIDI panner

Crossfader

The opposite of a pan control, a reverse panner if you like, is a crossfader. When you want to smoothly transfer between two sound sources by mixing them to a common signal path, the patch shown in Fig. 7.12 can be used. There are three signal inlets, two of them are signals to be mixed and one is a control signal to set the ratio (of course a message domain version would work equally well with appropriate anti-zipper smoothing). It can be used in the final stage of a reverb effects unit to set the wet/dry proportion, or in a DJ console to crossfade between two tunes. Just like the simple panner, the control signal is split into two parts, a direct version and the complement with each modulating an input signal. The output is the sum of both multipliers. This type is a linear

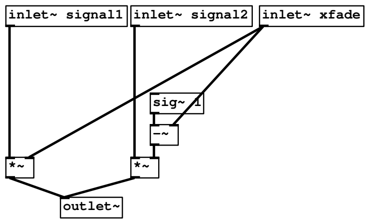


fig 7.12: crossfader

crossfader, but in some situations crossfades may be better with constant power fading done using sine or square root transfer functions.

Demultiplexer

A demultiplexer or signal source selector is a multi-way switch that can choose between a number of signal sources. Fig. 7.13 is useful in synthesiser construction where you want to select from a few different waveforms. In this design the choice is exclusive so only one input channel can be sent to the output at any time. A number at the control inlet causes `select` to choose one of four possible messages to send to `unpack`. The first turns off all channels, the second switches on only channel one and so forth. The Boolean values appearing in the `unpack` output are converted to signals and then lowpassed at 80Hz to give a fast but click free transition.

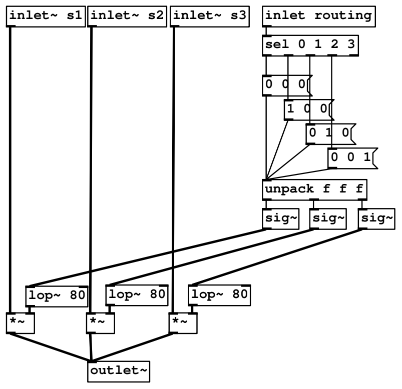


fig 7.13: demultiplex

SECTION 7.2

Audio file tools

Monophonic sampler

A useful object to have around is a simple sampler that can grab a few seconds of audio input and play it back. Audio arrives at the first inlet and is scaled by a gain stage before being fed to `tabwrite~`. It's nice to have a gain control so that you don't have to mess with the level of the signal you are recording elsewhere. In Fig. 7.14 a table of 88200 samples is created called `$0-a1`, so we have a couple of seconds recording time. Obviously this can be changed in the code or a control created to use the `resize` method. When it receives a bang `tabwrite~` starts recording from the beginning of the table. To play back the recording we issue a bang to `tabplay~` which connects directly to the outlet. The use of dollar arguments means this patch can be abstracted and multiple instances created, it's not unusual to want a bunch of samplers when working on a sound.

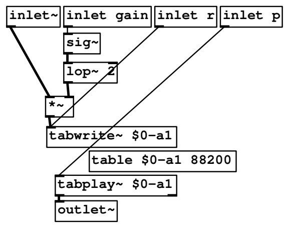


fig 7.14: simple sampler

Using the sampler is very easy. Create an instance and connect it to a signal source via the first inlet. In Fig. 7.15 the left audio input is taken from `adc~`. A slider with a range 0.0 to 1.0 connects to the gain inlet and two bang buttons are used to start recording or playback. Sound files of up to 3min can be stored happily in memory. Beyond this limit you need to use other objects for 32 bit machines because the sound quality will suffer due to pointer inaccuracies. If you have files longer than 3 minutes then you may want to think about using disk based storage and playback.

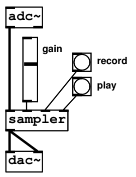


fig 7.15: using a sampler

File recorder

When creating sounds for use in other applications, like multitracks or samplers you could choose to record the output of Pd directly from the `dac~` using your favourite wave file editor or software like *Timemachine*. This could mean editing long recordings later, so sometimes you want to just write fixed length files directly from Pd.

In Fig. 7.16 we see a file writer in use, which I will show you how to make in a moment. It catches audio, perhaps from other patches, on a bus called `audio`. It was created with two arguments, the length of each file to record (in this case 1s) and the name of an existing folder beneath the current working directory in which to put them. Each time you hit the `start` button a new file is written to disk and then the `done` indicator tells you when it's finished. A numerical suffix is appended to each file, which you can see on the second outlet, in order to keep track of how many files you've created. The internals of the file writer are shown

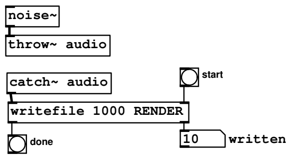


fig 7.16: Using a file writer

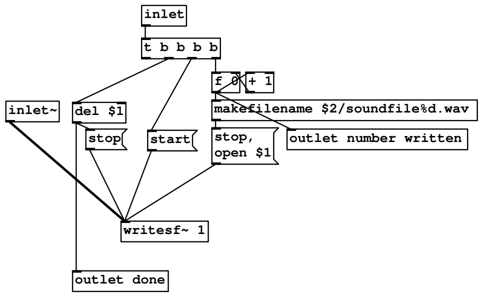


fig 7.17: Making a file writer

in Fig. 7.17. Audio comes into the first inlet and to the `writesf~` object which has an argument of 1, so writes a single channel (mono) file. There are three commands that `writesf~` needs, the name of a file to open for writing, a start command, and a stop command. Each bang on the second inlet increments a counter and the value of this is appended to the current file name using makefilename which can substitute numerical values into a string like the C printf statement does. This string is then substituted after the `open` keyword in the following message. As soon as this is done a `start` message is sent to writesf~ and a bang to the delay which waits for a period given by the first argument before stopping writesf~.

Loop player

A looping sample player is useful in many situations, to create a texture from looped background samples, or to provide a beat from a drum loop, especially if you need a continuous sound to test some process with. In Fig. 7.18 we see a patch that should be created as an abstraction so that many can be instantiated if required. It's operation is unsophisticated, just playing a loop of a sound file forever. When the abstraction receives a bang openpanel is activated and provides a nice file dialogue for you to choose a sound file. You should pick a Microsoft .wav or Mac .aiff type, either stereo or mono will do but this player patch will only give mono output. The name and path of this file is passed through the trigger "any" outlet and packed as the first part of a list along with a second part which is a symbol `$0-a`. The second symbol is the name of our storage table, the place in memory where the contents of the soundfile will be put once read. It has the prefix `$-` to give it local scope so we can have many sample loop players in a patch. Now the elements of the list will be substituted in $1 and $2 of the message `read -resize $1 $2`, which forms a complete command to soundfiler telling it to read in a sound file and put it in an array resizing the array as required. Once this operation is complete soundfiler returns the number of bytes read, which in this case we ignore and simply trigger a new bang message to start tabplay~. Notice the argument is the name of the array living in the table just above it. tabplay~ will now play once through the file at its original sample rate, so there is no need to tune it. When it has finished, the right outlet emits a bang. We take this bang, buffering it through another trigger and apply it back to the tabplay~ inlet, which means it plays the sound forever in a loop. A zero arriving at the second inlet allows you to stop the loop playing.

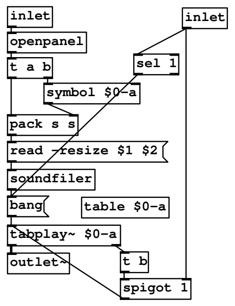


fig 7.18: sample loop player

SECTION 7.3

Events and sequencing

Now let's look at a few concepts used for creating time, sequences and event triggers.

Timebase

At the heart of many audio scenes or musical constructions is a timebase to drive events. We've already seen how to construct a simple timebase from a metronome and counter. A more useful timebase is given in Fig. 7.19 that allows you to specify the tempo as beats per minute (BPM) and to add a “swing”[1] to the beat. Notice first that start and stop control via the first inlet also resets

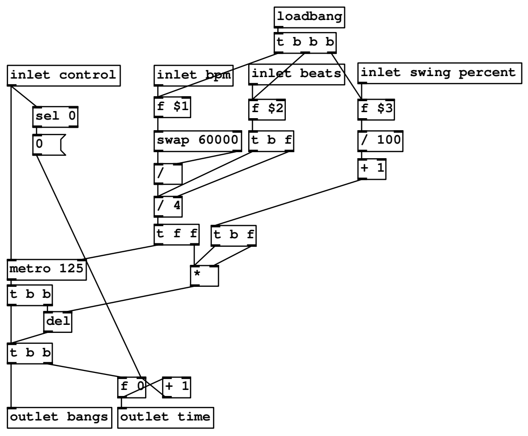


fig 7.19: A more useful musical timebase abstraction with BPM and swing

the counter when the timebase is stopped. Bangs from metro are duplicated with a delay object so we can position every other beat relative to the main rhythm. To convert beats per minute to a period in milliseconds it is divided by 60000 and multiplied by the number of beats per bar. The last parameter provides swing as a percentage which is added to the delay prior to incrementing the counter.

Select sequencer

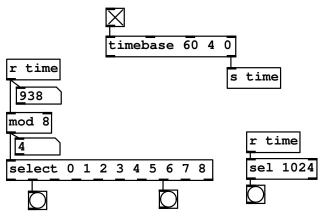


fig 7.20: Select based triggering

The simplest way to obtain regular patterns for repetitive sounds is by using mod to wrap the incoming time to a small range, say 8 beats, and then use select to trigger events within this range. You do not have to fill out all the select values, so for example, to produce a single trigger at *time* = 1024 you can connect one select matching this number. A good practice is to broadcast a global time message so that other patches can pick up a common reference. In Fig. 7.20 the output from the timebase abstraction goes to a send. To create a sequencer where you can manually set the time at which an event is triggered, use a com-

[1]Swing is where every other beat is moved slightly in time giving a different feel to the rhythm.

bination of == and select with a number box attached to the cold inlet of == and the current time going to the left inlet.

Partitioning time

For long musical compositions, interactive installations or generating event structures for a long game or animation you may want to offset timing sequences by a large number but keep the relative timings within a section. This is how bars and measures work in a traditional sequencer. In Fig. 7.21 moses is used to split the global time into smaller frames of reference. A chain of moses objects splits off numbers that fall within a range. You can see that the last value present on the left outlet of the first split was 127. Numbers of 128 or more are passing through the right outlet and into the second moses which partitions values between 128 and 255. We subtract the base value of 128 from this stream to reposition it, as if it were a sequence starting at zero. This can be further processed, such as wrapping it into the range 0 to 64 to create 2 bars of 64 beats in the range 128 to 256. In Fig. 7.21 you see the timebase at 208, which is in the second bar of the partitioned timeframe.

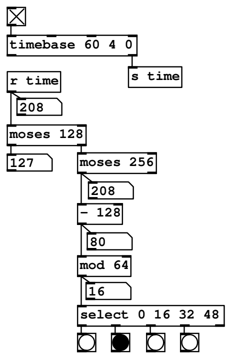


fig 7.21: Bar offset by partitioning time

Dividing time

With time expressed as a number you can perform arithmetic on it to obtain different rates. Be aware that although the value of numerical time changes with a different scale it still updates at the rate set by the timebase. Since for musical purposes you want to express time in whole beats and bars a problem is presented. Dividing time by two and rounding it to an integer means two messages will now be sent with the same value. To get around this problem change is used so that redundant messages are eliminated. Using int means values are rounded to the time floor, so if rhythms constructed this way seem one beat out of alignment you can try using a "closest integer" rounding explained earlier. Sometimes rounding time is not what you want as shown in the next example.

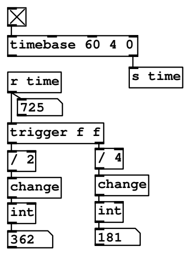


fig 7.22: Dividing time into different rates

Event synchronised LFO

An application and pitfall of timebase division is shown in Fig. 7.23 where low frequency control signals are derived from the timebase. Notice how the sine wave is not interpolated, so you get two or four consecutive equal values when using a divided timebase. This makes the LFO jumpy so to avoid it we scale the raw time values before the trig operation using a higher mod and /. It

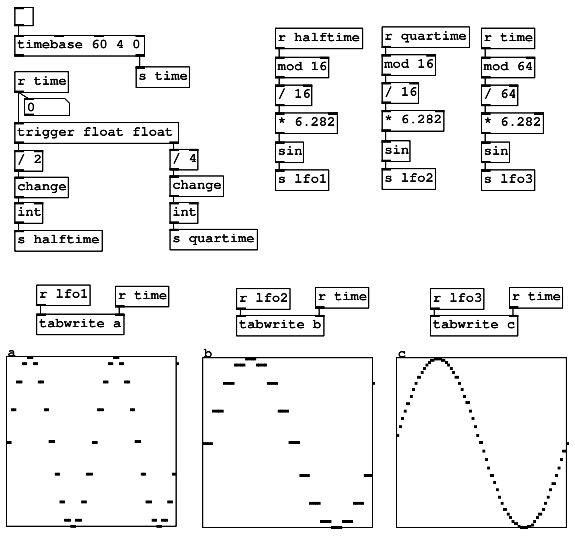


fig 7.23: Synchronous message LFOs

illustrates why you should often use a timebase that is a large multiple (say 64 times) of the real event rate you want. You might use this to create interesting polyrhythms or elaborate slow moving control signals for wind, rain or spinning objects.

List sequencer

An alternative to an absolute timebase is using lists and delays to make a relative time sequencer. Events are stored in a list, which we define to have a particular meaning to a sequencer that will interpret it. In this case the list is read in pairs, an event type and a time offset from the last event. So, a list like {*1 0 2 200 1 400*} describes three events and two event types. Event 1 occurs at *time* = 0 and then at *time* = 200 event 2 occurs, followed by event 1 again at *time* = 200 + 400 = 600. Times are in milliseconds and event types usually correspond to an object name or a MIDI note number. The patch in Fig. 7.24 is hard to follow, so I will describe it in detail. The sequence list arrives at the first inlet of `list split 2` where it is chopped at the second element. The first two elements pass to the `unpack` where they are separated and processed, while the remainder of the list passes out of the second outlet of `list split 2` and into the right inlet of `list append`. Returning to `unpack`, our first half of the current pair which identifies a float event type is sent to the cold inlet of a `float` where it waits, while the second part which represents a time delay is passed to `delay`. After a delay corresponding to this second value `delay` emits a bang message which flushes out the value stored in `float` for output. Finally, `list append` is

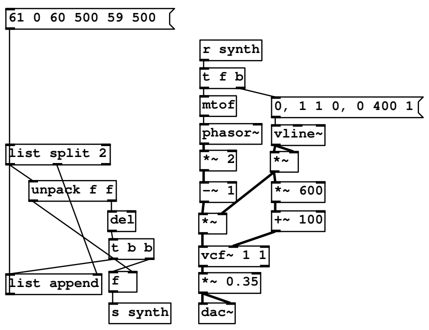


fig 7.24: An asynchronous list sequencer

banged so the remainder of the list is passed back to `list split 2` and the whole process repeats, chomping 2 elements off each time until the list is empty. To the right of Fig. 7.24 is a simple monophonic music synthesiser used to test the sequencer. It converts MIDI note numbers to Hertz with `mtof` and provides a filtered sawtooth wave with a 400ms curved decay envelope. To scale the sequence delay times, and thus change the tempo without rewriting the entire list, you can make each time offset be a scaling factor for the delay which is then multiplied by some other fraction. List sequencers of this type behave asynchronously, so don't need a timebase.

Textfile control

Eventually lists stored in message boxes become unwieldy for large data sets and it's time to move to secondary storage with textfiles. The `textfile` object provides an easy way to write and read plain text files. These can have any format you like, but a general method is to use a comma or linebreak delimited structure to store events or program data. It is somewhat beyond this textbook to describe the many ways you can use this object, so I will present only one example of how to implement a text file based MIDI sequencer. A combination of `textfile` and `route` can provide complex score control for music or games. If you need even larger data sets with rapid access a SQL object is available in `pd-extended` which can interface to a database.

Starting at the top left corner of Fig. 7.25 you can see a monophonic synthesiser used to test the patch. Replace this with a MIDI note out function if you like. The remainder of the patch consists of two sections, one to store and write the sequence and one to read and play it back. Recording commences when the `start-record` button is pressed. This causes a `clear` message to be sent to `textfile`, the list accumulator is cleared and the `timer` object reset. When a note is received by `notein` and then reduced to just its note-on value by `stripnote` it passes to the trigger unit below which dispatches two bangs to `timer`. The result of this is for `timer` to output the time since the last bang it received, then

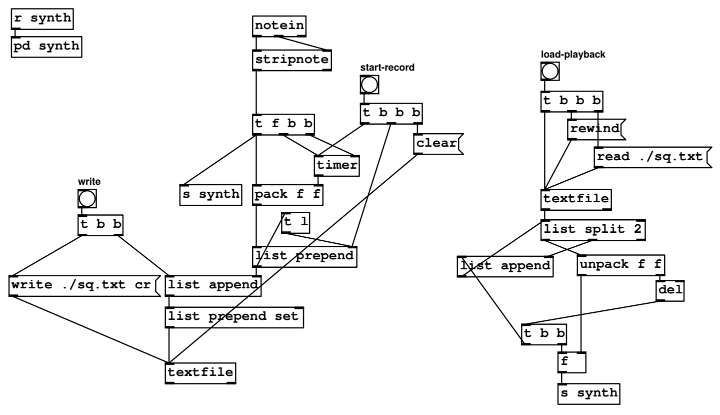


fig 7.25: A MIDI sequencer that uses textfiles to store data

restart from zero. This time value, along with the current MIDI note number, is packed by `pack` into a pair and appended to the list accumulator. When you are done playing, hit the `write` button to flush the list into `textfile` and write it to a file called `sq.txt` in the current working directory. Moving to the load and replay side of things, banging the `load-replay` button reads in the textfile and issues a `rewind` message setting `textfile` to the start of the sequence. It then receives a bang which squirts the whole list into a list sequencer like the one we just looked at.

SECTION 7.4

Effects

For the last part of this chapter I am going to introduce simple effects. Chorus and reverb are used to add depth and space to a sound. They are particularly useful in music making, but also have utility in game sound effects to thicken up weaker sources. Always use them sparingly and be aware that it is probably better to make use of effects available in your external mixer, as plugins, or as part of the game audio engine.

Stereo chorus/flanger effect

The effect of chorus is to produce a multitude of sources by doubling up many copies of the same sound. To do this we use a several delays and position them slightly apart. The aim is to deliberately cause beating and swirling as the copies move in and out of phase with one another. In Fig. 7.26 an input signal at the first inlet is split three ways. An attenuated copy is fed directly to the right stereo outlet while two other copies are fed to separate delay lines. In the centre you see two variable delay taps, `vd~`, which are summed.

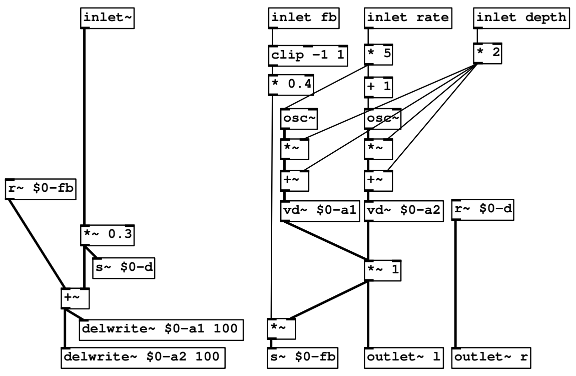


fig 7.26: A chorus type effect

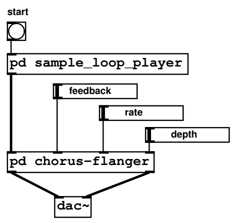


fig 7.27: Testing the chorus

A small part, scaled by the feedback value on the second inlet, is sent back to be mixed in with the input signal, while another copy is sent to the left stereo outlet. So there is a dry copy of the signal on one side of the stereo image and two time shifted copies on the other. By slowly varying the delay times with a couple of signal rate LFOs a swirling chorus effect is achieved. The low frequency oscillators are always 1Hz apart and vary between 1Hz and 5Hz. It is necessary to limit the feedback control to be sure the effect cannot become unstable. Notice that feedback can be applied in positive or negative phase to create a notching effect (phaser/flanger) and a reinforcing effect (chorus). Testing out the effect is best with a sample loop player. Try loading a few drum loops or music loop clips.

Simple reverberation

A reverb simulates dense reflections as a sound bounces around inside some space. There are several ways of achieving this effect, such as convolving a sound with the impulse response of a room or using allpass filters to do a similar thing. In Fig. 7.28 you can see a design for a recirculating reverb type that uses only delay lines. There are four delays which mutually feed back into one another, so once a signal is introduced into the patch it will circulate through a complex path. So that reinforcement doesn't make the signal level keep growing some feedback paths are negative. The recirculating design is known as a Schroeder reverb (this example by Claude Heiland-Allen) and mimics four walls of a room. As you can see the number of feedback paths gets hard to patch if we move to 6 walls (with floor and ceiling) or to more complex room shapes. Reverb design is a fine art. Choosing the exact feedback and delay values is not easy. If they are wrong then a feedback path may exist for certain frequencies producing an unstable effect. This can be hard to detect in practice and complex to predict in

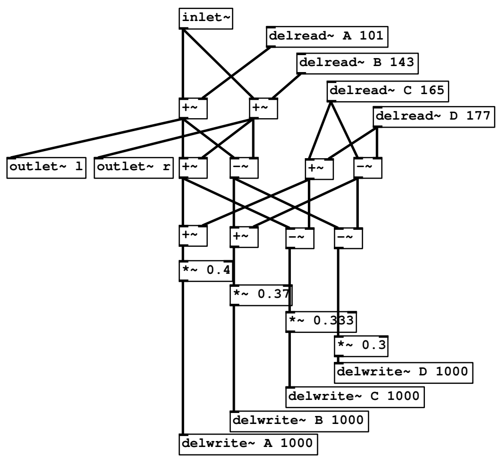


fig 7.28: A recirculating Schroeder reverb effect

theory. An apparently well designed reverb can mysteriously explode after many seconds or even minutes so a common design safety measure is to attenuate the feedback paths as the reverb decays away. What defines the reverb time is the point at which the reverb is has fallen to −60dB of the first reflection intensity. A good design should not be too coloured, which means feedback paths must not be too short leading to a pitched effect. The minimum delay time should be at least a quarter of the reverberation time and the lengths of delays should be prime, or collectively co-prime [2]. The density of the reverb is important too. Too little and you will hear individual echos, too much and the effect will become muddy and noisy. Schroeder suggests 1000 echoes per second for a reasonable reverb effect. If you look in the `extra` directory that comes with Pd there are three nice reverb abstractions `rev1~`, `rev2~` and `rev3~` by Miller Puckette

Exercises

Exercise 1

Create any one of the following effects.

- Guitar tremolo effect
- Multi-stage phaser
- Multi-tap tempo-sync delay
- A high quality vocal reverb

Exercise 2

Create a sequencer that provides any two of,

[2]A set of integers with no common factors are said to be collectively co-prime.

- Hierarchical structure
- Microtonal tuning scales
- Polyrhythmic capabilities
- A way to load and save your compositions

Exercise 3

Design and implement a mixing desk with at least three of,

- MIDI or OSC parameter automation
- Switchable fader and pan laws
- Surround sound panning (eg 5.1, quadraphonic)
- Effect send and return bus
- Accurate signal level monitoring
- Group buses and mute groups
- Scene store and recall

Exercise 4

Essay: Research datastructures in Pd. How can graphical representations help composition? What are the limitations of graphics in Pd? Generally, what are the challenges for expressing music and sound signals visually?

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Online resources

http://puredata.info/ is the site of the main Pure Data portal.

http://crca.ucsd.edu/ is the current home of official Pure Data documentation by Miller Puckette.

Beau Sievers "The Amateur Gentleman's Introduction to Music Synthesis" An introductory online resource geared toward synth building in Pure Data. http://beausievers.com/synth/synthbasics/

http://www.musicdsp.org/ is the home of the music DSP list archive, with categorised source code and comments.

http://www.dafx.de/ is home of the DAFx (Digital Audio Effects) project containing many resources.

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