An Instrument Design TOOTorial

Richard Boulanger

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Introduction

Csound instruments are created in an *orchestra* file, and the list of notes to play is written in a separate *score* file. Both are created using a standard word processor. When you run Csound on a specific orchestra and score, the score is sorted and ordered in time, the orchestra is translated and loaded, the wavetables are computed and filled, and then the score is performed. The score drives the orchestra by telling the specific instruments when and for how long to play, and what parameters to use during the course of each note event.

Unlike today's commercial hardware synthesizers, which have a limited set of oscillators, envelope generators, filters, and a fixed number of ways in which these can be interconnected, Csound's power is not limited. If you want an instrument with hundreds of oscillators, envelope generators, and filters you just type them in. More important is the freedom to interconnect the modules, and to interrelate the parameters which control them. Like acoustic instruments, Csound instruments can exhibit a sensitivity to the musical context, and display a level of “musical intelligence” to which hardware synthesizers can only aspire.

Because the intent of this tutorial is to familiarize the novice with the syntax of the language, we will design several simple instruments. You will find many instruments of the sophistication described above in **The Csound Book** or by following the links at **Csounds.com**. A study of

these will reveal Csound's real power. Still, hopefully you will find in these simple examples the building block you require and I encourage you to modify and extend each of them and recommend that you try to make each score more involved and more musical. Well, let's get started!

In Csound, the *orchestra file* has two main parts:

- *the header section* - defining the sample rate, control rate, and number of output channels.
- *the instrument section* - in which the instruments are designed.

The Header Section

A Csound orchestra generates signals at two rates - an audio sample rate and a control sample rate. Each can represent signals with frequencies no higher than half that rate, but the distinction between audio signals and sub-audio control signals is useful since it allows slower moving signals to require less compute time. In the header below, we have specified a sample rate of 44.1 kHz, a control rate of 4410 Hz, and then calculated the number of samples in each control period using the formula: ksmps = sr / kr

```
sr = 44100
kr = 4410
ksmps = 10
nchnls = 1
```

In Csound orchestras and scores, spacing is arbitrary. It is important to be consistent in laying out your files, and you can use spaces to help this. In the Tutorial Instruments shown below you will see we have adopted one convention. The reader can choose his or her own.

The Instrument Section

All instruments are numbered and are referenced thus in the score. Csound instruments are similar to patches on a hardware synthesizer. Each instrument consists of a set of “unit generators,” or

software "modules," which are "patched" together with "i/o" blocks — i-, k-, or a-rate variables. Unlike a hardware module, a software module has a number of variable "arguments" which the user sets to determine its behavior. The four types of variables are:

- setup only
- i-rate variables, changed at the note rate
- k-rate variables, changed at the control signal rate
- a-rate variables, changed at the audio signal rate

Orchestra Statements

Each statement occupies a single line and has the same basic format:

```
result action arguments
```

To include an oscillator in our orchestra, you might specify it as follows:

```
a1 oscil 10000, 440, 1
```

The three "arguments" for this oscillator set its amplitude (10000), its frequency (440Hz), and its wave shape (1). The output is put in i/o block *a1*. This output symbol is significant in prescribing the rate at which the oscillator should generate output — here the audio rate. We could have named the result anything (e.g. *asig*) as long as it began with the letter "a".

Comments

To include text in the orchestra or score which will not be interpreted by the program, precede it with a semicolon. This allows you to fully comment your code. On each line, any text which follows a semicolon will be ignored by the orchestra and score translators.

Toot 1: Play One Note

The first orchestra file, called *Toot01.orc* contains a single instrument which uses an **oscil** unit to play a 440Hz sine wave (defined by f1 in the score) at an amplitude of 10000.

```
        instr 1
a1      oscil   10000, 440, 1
        out     a1
        endin
```

Toot01.orc

Run this with its corresponding score file, *toot1.sco*

```
f1  0   4096    10 1  ; use GEN10 to compute a sine wave

;ins    strt    dur
i1      0       4

e                     ; indicates the end of the score
```

Toot01.sco

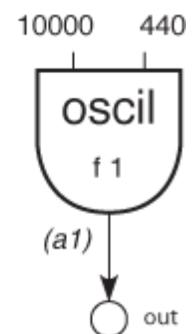


Figure 1 Block diagram of instr 1.

Toot 2: P-Fields

The first instrument was not interesting because it could play only one note at one amplitude level. We can make things more interesting by allowing the pitch and amplitude to be defined by parameters in the score. Each column in the score constitutes a parameter field, numbered from the left. The first three parameter fields of the **i statement** have a reserved function:

```
p1 = instrument number
p2 = start time
p3 = duration
```

All other parameter fields are determined by the way the sound designer defines his instrument. In the instrument below, the oscillator's amplitude argument is replaced by p4 and the frequency argument by p5. Now we can change these values at i-time, i.e. with each note in the score. The orchestra and score files now look like:

```
          instr 2
a1        oscil     p4, p5, 1      ; p4=amp
          out       a1             ; p5=freq
          endin
```

Toot02.orc

```
f1   0    4096 10 1      ; sine wave

;ins strt dur  amp(p4)   freq(p5)
i2   0    1    2000      880
i2   1.5  1    4000      440
i2   3    1    8000      220
i2   4.5  1    16000     110
i2   6    1    32000     55

e
```

Toot02.sco

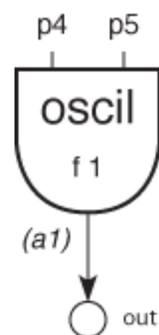


Figure 2 Block diagram of instr 2.

Toot 3: Envelopes

Although in the second instrument we could control and vary the overall amplitude from note to note, it would be more musical if we could contour the loudness during the course of each note. To do this we'll need to employ an additional unit generator **linen**, which the Csound reference manual defines as follows:

```
kr linen kamp, irise, idur, idec
ar linen xamp, irise, idur, idec
```

linen is a signal modifier, capable of computing its output at either control or audio rates. Since we plan to use it to modify the amplitude envelope of the oscillator, we'll choose the latter version. Three of linen's arguments expect **i**-rate variables. The fourth expects in one instance a k-rate variable (or anything slower), and in the other an x-variable (meaning a-rate or anything slower). Our **linen** we will get its amp from p4.

The output of the **linen** (*k1*) is patched into the *kamp* argument of an **oscil**. This applies an envelope to the **oscil**. The orchestra and score files now appear as:

```
          instr 3                        ; p3=duration of note
```

```
k1         linen     p4, p6, p3, p7      ; p4=amp
a1         oscil     k1, p5, 1           ; p5=freq
           out       a1                  ; p6=attack time
           endin                         ; p7=release time
```

toot03.orc

```
f1   0    4096 10 1      ; sine wave

;ins strt dur  amp(p4)    freq(p5)  attack(p6)     release(p7)
i3   0    1    10000      440       0.5            0.7
i3   1.5  1    10000      440       0.9            0.1
i3   3    1    5000       880       0.02           0.99
i3   4.5  1    5000       880       0.7            0.01
i3   6    2    20000      220       0.5            0.5

e
```

toot03.sco

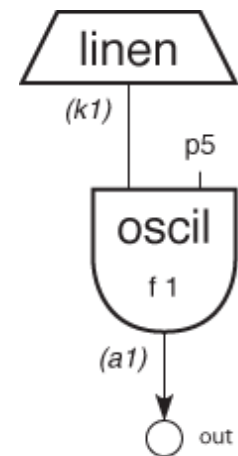


Figure 3 Block diagram of instr 3.

Toot 4: Chorusing

Next we'll animate the basic sound by mixing it with two slightly de-tuned copies of itself. We'll employ Csound's **cpspch** value converter which will allow us to specify the pitches by octave and pitch-class rather than by frequency, and we'll use the **ampdb** converter to specify loudness in dB rather than linearly.

Since we are adding the outputs of three oscillators, each with the same amplitude envelope, we'll scale the amplitude before we mix them. Both *iscale* and *inote* are arbitrary names to make the design a bit easier to read. Each is an i-rate variable, evaluated when the instrument is initialized.

```
          instr 4
iamp      =         ampdb(p4)           ; convert decibels to linear amp
iscale    =         iamp * .333         ; scale the amp at initialization
inote     =         cpspch(p5)          ; convert octave.pitch to cps

k1        linen     iscale, p6, p3, p7  ; p4=amp

a3        oscil     k1, inote*.996, 1   ; p5=freq
a2        oscil     k1, inote*1.004, 1  ; p6=attack time
a1        oscil     k1, inote, 1        ; p7=release time

a1        =         a1+a2+a3
          out       a1
          endin
```

toot04.orc

```
f1   0    4096 10 1      ; sine wave

;ins strt dur  amp  freq      attack    release
i4   0    1    75   8.04      0.1       0.7
i4   1    1    70   8.02      0.07      0.6
i4   2    1    75   8.00      0.05      0.5
i4   3    1    70   8.02      0.05      0.4
i4   4    1    85   8.04      0.1       0.5
```

```
i4   5    1    80   8.04      0.05      0.5
i4   6    2    90   8.04      0.03      1.
```

toot04.sco

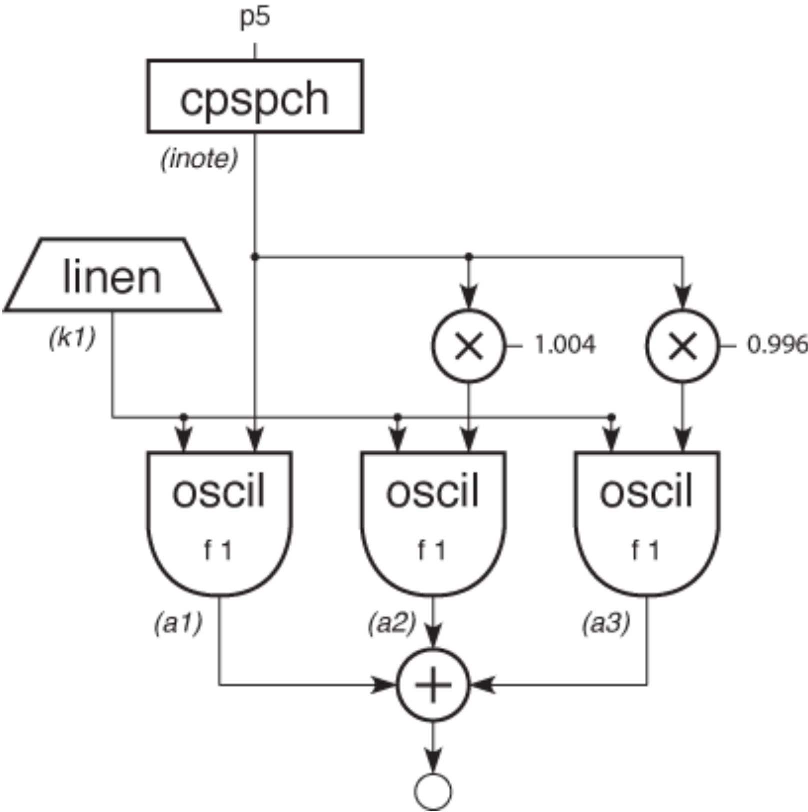


Figure 4 Block diagram of instr 4.

Toot 5: Vibrato

To add some delayed vibrato to our chorusing instrument we use another oscillator for the vibrato

and a line segment generator, **linseg**, as a means of controlling the delay. **linseg** is a k-rate or a-rate signal generator which traces a series of straight line segments between any number of specified points. The Csound manual describes it as:

```
kr linseg ia, idur1, ib[, idur2, ic[...]]
ar linseg ia, idur1, ib[, idur2, ic[...]]
```

Since we intend to use this to slowly scale the amount of signal coming from our vibrato oscillator, we'll choose the k-rate version. The i-rate variables: *ia*, *ib*, *ic*, etc., are the values for the points. The i-rate variables: *idur1*, *idur2*, *idur3*, etc., set the duration, in seconds, between segments.

```
          instr 5
irel      =         0.01                                ; set vibrato release time
idel1     =         p3 * p10                            ; calculate initial delay (% of dur)
isus      =         p3 - (idel1 + irel)                 ; calculate remaining duration

iamp      =         ampdb(p4)
iscale    =         iamp * .333                         ; p4=amp
inote     =         cpspch(p5)                          ; p5=freq

k3        linseg    0, idel1, p9, isus, p9, irel, 0     ; p6=attack time
k2        oscil     k3, p8, 1                           ; p7=release time
k1        linen     iscale, p6, p3, p7                  ; p8=vib rate

a3        oscil     k1, inote*.995+k2, 1                ; p9=vib depth
a2        oscil     k1, inote*1.005+k2, 1               ; p10=vib delay (0-1)
a1        oscil     k1, inote+k2, 1

          out       a1+a2+a3
          endin
```

toot05.orc

```
f1   0    4096 10 1      ; sine wave

;ins strt dur  amp  freq      atk  rel  vibrt     vbdpt     vbdel
i5   0    3    86   10.00     0.1  0.7  7         6         .4
```

```
i5   4    3    86   10.02     1    0.2  6           6          .4
i5   8    4    86   10.04     2    1    5           6          .4
```

toot05.sco

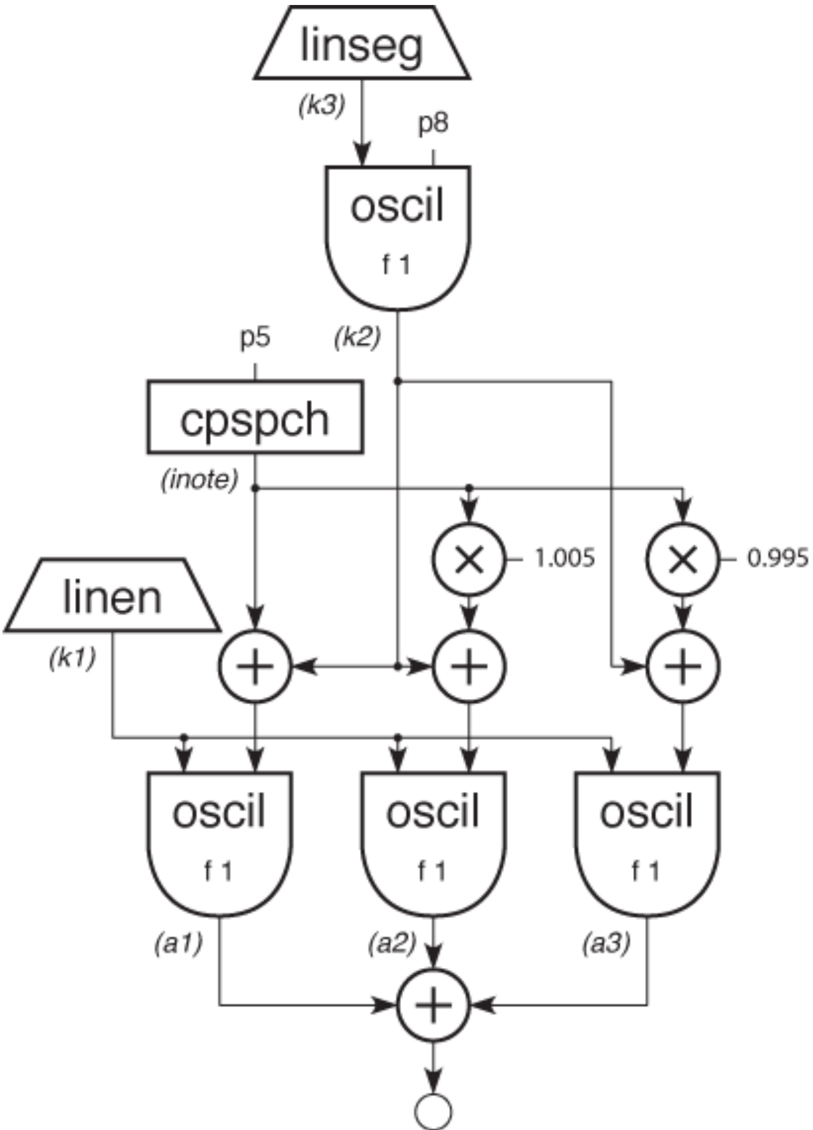


Figure 5 Block diagram of instr 5.

Toot 6: GENs

The first character in a score statement is an **opcode**, determining an action request; the remaining data consists of numeric parameter fields (p-fields) to be used by that action. So far we have been dealing with two different opcodes in our score: f and i. i statements, or note statements, invoke the p1 instrument at time p2 and turn it off after p3 seconds; all remaining p-fields are passed to the instrument.

On the other hand, **f statement**s, or lines with an opcode of **f**, invoke function-drawing subroutines called **GENS**. In Csound there are currently twenty-three GEN routines which fill wavetables in a variety of ways. For example, **GEN01** transfers data from a soundfile; **GEN07** allows you to construct functions from segments of straight lines; and **GEN10**, which we've been using in our scores so far, generates composite waveforms made up of a weighted sum of simple sinusoids. We have named the function “f1,” invoked it at time 0, defined it to contain 512 points, and instructed **GEN10** to fill that wavetable with a single sinusoid whose amplitude is 1. **GEN10** can in fact be used to *approximate* a variety of other waveforms, as illustrated by the following:

```
f1   0     2048 10   1                                                        ; Sine
f2   0     2048 10   1    0.5  0.3  0.25 0.2  0.167     0.14      0.125     .111      ; Sawtooth
f3   0     2048 10   1    0    0.3  0    0.2  0         0.14      0         .111      ; Square
f4   0     2048 10   1    1    1    1    0.7  0.5       0.3       0.1                 ; Pulse
```

For the opcode f, the first four p-fields are interpreted as follows:

- p1 - table number - In the orchestra, you reference this table by its number.
- p2 - creation time - The time at which the function is generated.
- p3 - table size - Number of points in table - must be a power of 2, or that plus 1.
- p4 - generating subroutine - Which of the 17 GENS will you employ.
- p5 - meaning determined by the particular GEN subroutine.

In the instrument and score below, we have added three additional functions to the score, and

modified the orchestra so that the instrument can call them via p11.

```
          instr 6
ifunc     =         p11                                ; select the basic waveform
irel      =         0.01                               ; set vibrato release time
idel1     =         p3 * p10                           ; calculate initial delay (% of dur)
isus      =         p3 - (idel1 + irel)                ; calculate remaining duration

iamp      =         ampdb(p4)
iscale    =         iamp * .333                        ; p4=amp
inote     =         cpspch(p5)                         ; p5=freq

k3        linseg    0, idel1, p9, isus, p9, irel, 0    ; p6=attack time
k2        oscil     k3, p8, 1                          ; p7=release time
k1        linen     iscale, p6, p3, p7                 ; p8=vib rate

a3        oscil     k1, inote*.999+k2, ifunc           ; p9=vib depth
a2        oscil     k1, inote*1.001+k2, ifunc          ; p10=vib delay (0-1)
a1        oscil     k1, inote+k2, ifunc

          out       a1+a2+a3
          endin
```

toot06.orc

```
f1   0    2048 10   1                                                                 ; Sine
f2   0    2048 10   1    0.5  0.3  0.25 0.2  0.167     0.14      0.125     .111       ; Sawtooth
f3   0    2048 10   1    0    0.3  0    0.2  0         0.14      0         .111       ; Square
f4   0    2048 10   1    1    1    1    0.7  0.5       0.3       0.1                  ; Pulse

;ins strt dur  amp       frq       atk       rel       vbrt vbdpt     vibdl     waveform
i6   0     2   86        8.00      .03       .7        6    9         .8        1
i6   3     2   86        8.02      .03       .7        6    9         .8        2
i6   6     2   86        8.04      .03       .7        6    9         .8        3
i6   9     3   86        8.05      .03       .7        6    9         .8
```

toot06.sco

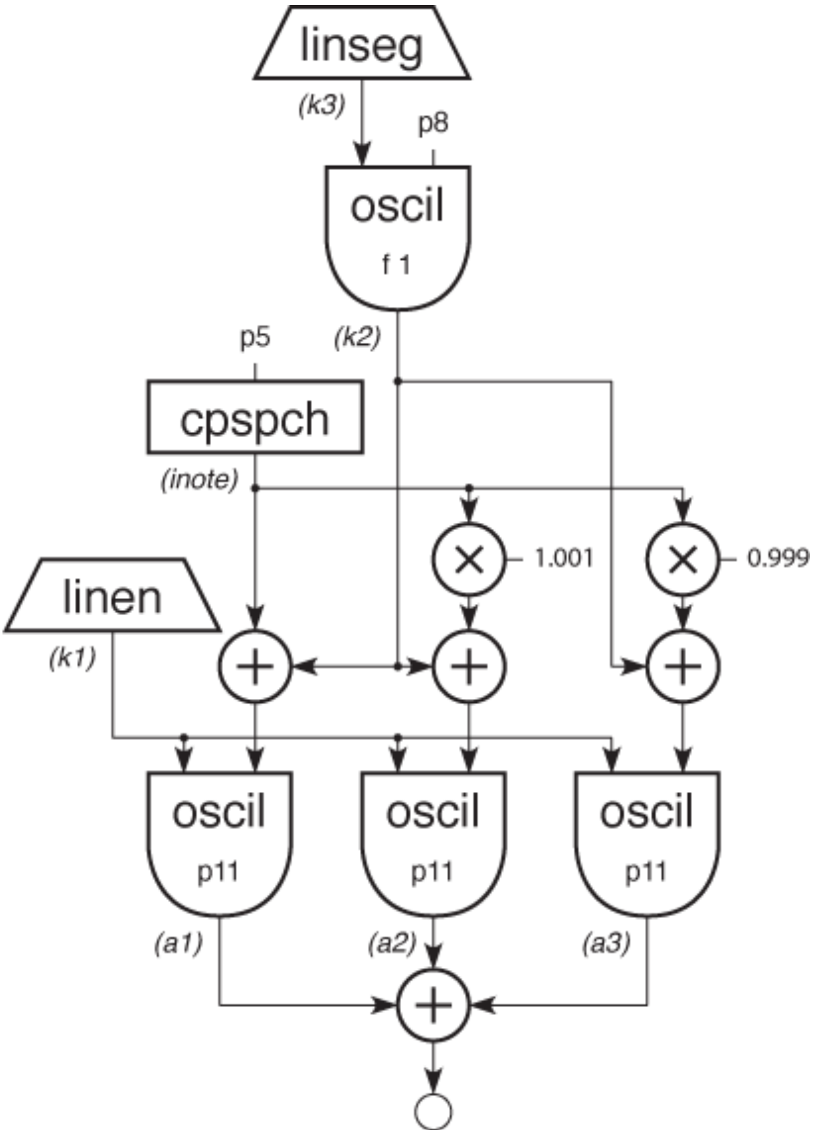


Figure 6 Block diagram of instr 6.

Toot 7: Crossfade

Now we will add the ability to do a linear crossfade between any two of our four basic waveforms. We will employ our delayed vibrato scheme to regulate the speed of the crossfade.

```
          instr 7
ifunc1    =         p11                                 ; initial waveform
ifunc2    =         p12                                 ; crossfade waveform

ifad1     =         p3 * p13                            ; calculate initial fade (% of dur)
ifad2     =         p3 - ifad1                          ; calculate remaining duration

irel      =         .01                                 ; set vibrato release time
idel1     =         p3 * p10                            ; calculate initial delay (% of dur)
isus      =         p3 - (idel1 + irel)                 ; calculate remaining duration

iamp      =         ampdb(p4)
iscale    =         iamp * .166                         ; p4=amp
inote     =         cpspch(p5)                          ; p5=freq

k3        linseg    0, idel1, p9, isus, p9, irel, 0     ; p6=attack time
k2        oscil     k3, p8, 1                           ; p7=release time
k1        linen     iscale, p6, p3, p7                  ; p8=vib rate
a6        oscil     k1, inote*.998+k2, ifunc2           ; p9=vib depth
a5        oscil     k1, inote*1.002+k2, ifunc2          ; p10=vib delay (0-1)
a4        oscil     k1, inote+k2, ifunc2                ; p11=initial wave
a3        oscil     k1, inote*.997+k2, ifunc1           ; p12=cross wave
a2        oscil     k1, inote*1.003+k2, ifunc1          ; p13=fade time
a1        oscil     k1, inote+k2, ifunc1

kfade     linseg    1, ifad1, 0, ifad2, 1
afunc1    =         kfade * (a1+a2+a3)
afunc2    =         (1 - kfade) * (a4+a5+a6)

          out       afunc1 + afunc2
          endin
```

toot07.orc

```
f1   0    2048 10   1                                               ; Sine
```

```
f2   0    2048 10   1    0.5  0.3  0.25 0.2  0.167     0.14      0.125     .111      ; Sawtooth
f3   0    2048 10   1    0    0.3  0    0.2  0         0.14      0         .111      ; Square
f4   0    2048 10   1    1    1    1    0.7  0.5       0.3       0.1                 ; Pulse

;in st    dur  amp  frq       atk  rel  vbrt vbdp vbdl stw  ndw  crstm
i7   0    5    96   8.07      .03  .1   5    6    .99  1    2    .1
i7   6    5    96   8.09      .03  .1   5    6    .99  1    3    .1
i7  12    8    96   8.07      .03  .1   5    6    .99  1    4    .
```

toot07.sco

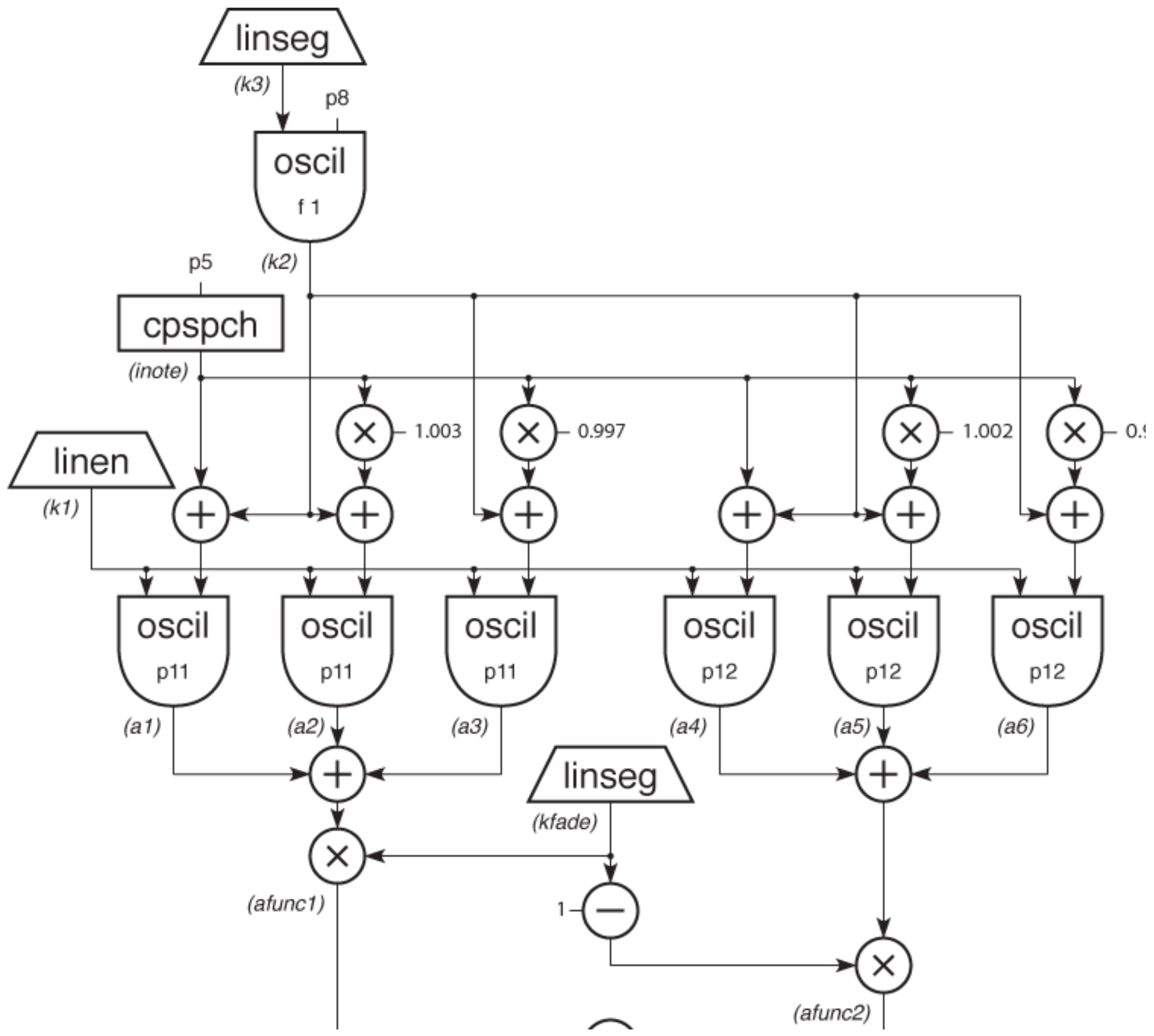

linseg
(k3)
p8
oscil
f 1
p5
(k2)
cpspch
(inote)
1.003
0.997
1.002
linen
(k1)
oscil
p11
oscil
p11
oscil
p11
oscil
p12
oscil
p12
oscil
p12
(a1)
(a2)
(a3)
(a4)
(a5)
(a6)
linseg
(kfade)
(afunc1)
1
(afunc2)

Figure 7 Block diagram of instr 7.

Toot 8: Soundin

Now instead of continuing to enhance the same instrument, we will design a totally different one. In it we'll read a soundfile into the orchestra, apply an amplitude envelope to it, and add some reverb. To do this we will employ Csound's **soundin** and **reverb** generators. The first is described as:

```
a1 soundin ifilcod[, iskiptime[, iformat]]
```

soundin derives its signal from a pre-existing file. *ifilcod* is either the filename in double quotes, or an integer suffix (.n) to the name “soundin”. Thus the file soundin.5 could be referenced either by the quoted name or by the integer 5. To read from 500ms into this file we might say:

```
a1 soundin “soundin.5”, .5
```

The Csound **reverb** generator is actually composed of four parallel **comb** filters plus two **alpass** filters in series. Although we could design a variant of our own using these same primitives, the preset reverb is convenient, and simulates a natural room response via internal parameter values. Only two arguments are required the input (*asig*) and the reverb time (*krvt*)

```
ar reverb asig, krvt
```

The soundfile instrument with artificial envelope and a reverb (included directly) is as follows:

```
          instr 8
idur      =         p3
iamp      =         p4
iskiptime =         p5
iattack   =         p6
irelease  =         p7
```

```
irvbtime  =         p8
irvbgain  =         p9

kamp      linen     iamp, iattack, idur, irelease
asig      soundin   "hellorcb.aif", iskiptime
arampsig  =         kamp * asig
aeffect   reverb    asig, irvbtime
arvbretrn =         aeffect * irvbgain
          out       arampsig + arvbretrn
          endin
```

toot08.orc

```
;ins strt dur  amp  skip atk  rel        rvbt rvbgain
i8   0    2.28 .3   0    .03  .1         1.5  .3
i8   4    1.6  .3   1.6  .1   .1         1.1  .4
i8   5.5  2.28 .3   0    .5   .1         2.1  .2
i8   6.5  2.28 .4   0    .01  .1         1.1  .1
i8   8    2.28 .5   0.1  .01  .1         0.1  .1
```

toot08.sco

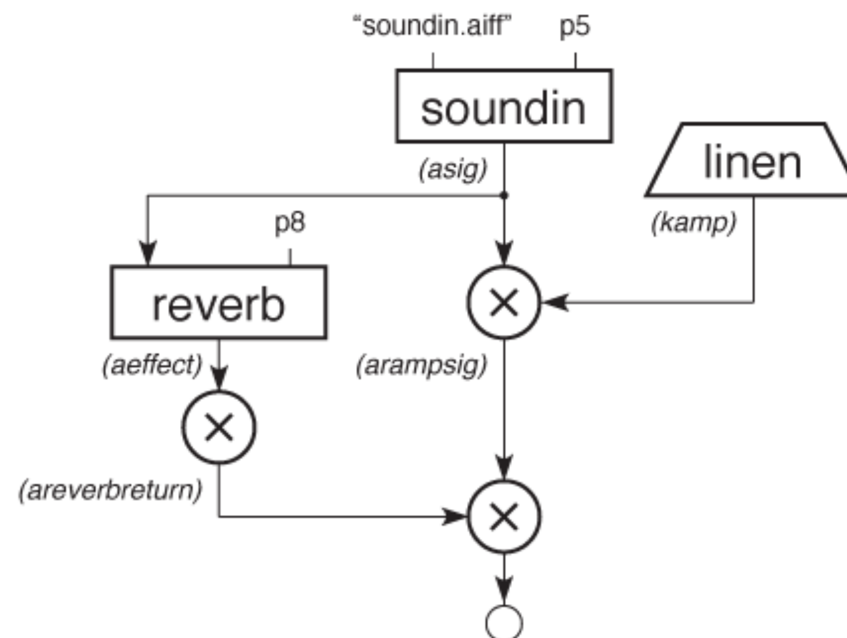


Figure 8 Block diagram of instr 8.

Toot 9: Global Stereo Reverb

In the previous example you may have noticed the soundin source being "cut off" at ends of notes, because the reverb was *inside*the instrument itself. It is better to create a companion instrument, a global reverb instrument, to which the source signal can be sent. Let's also make this stereo.

Variables are named cells which store numbers. In Csound, they can be either *local* or *global*, are available continuously, and can be updated at one of four rates - setup, i-rate, k-rate, or a-rate.

Local variables (which begin with the letters p, i, k, or a) are private to a particular instrument. They cannot be read from, or written to, by any other instrument.

Global Variables are cells which are accessible by all instruments. Three of the same four variable types are supported (i, k, and a), but these letters are preceded by the letter "g" to identify

them as "global." Global variables are used for "broadcasting" general values, for communicating between instruments, and for sending sound from one instrument to another.

The reverb *instr 99* below receives input from *instr 9* via the global a-rate variable *garvbsig*. Since *instr 9* adds into this global, several copies of *instr 9* can do this without losing any data. The addition requires *garvbsig* to be cleared before each k-rate pass through any active instruments. This is accomplished first with an **init** statement in the orchestra header, giving the reverb instrument a higher number than any other (instruments are performed in numerical order), and then clearing *garvbsig* within *instr 99* once its data has been placed into the reverb.

```
          instr 9
idur      =         p3
iamp      =         p4
iskiptime =         p5
iattack   =         p6
irelease  =         p7
ibalance  =         p8                         ; 1 = left, .5 = center, 0 = right
irvbgain  =         p9

kamp      linen     iamp, iattack, idur, irelease
asig      soundin   "hellorcb.aif", iskiptime
arampsig  =         kamp * asig
          outs      arampsig * ibalance, arampsig * (1 - ibalance)
garvbsig  =         garvbsig + arampsig * irvbgain
          endin


          instr 99
irvbtime  =         p4
asig      reverb    garvbsig,  irvbtime        ; put global sig into reverb
          outs      asig, asig
garvbsig  =         0                          ; then clear it
          endin
```

toot09.orc

```
;ins strt dur  rvbtime
i99  0    10   2.2
```

```
;ins strt dur  amp  skip atk  rel  blnce(0-1)     rvbsend
i9   0    1.2  .5   0    .02  .1   1              .2
i9   2    1.4  .5   0    .03  .1   0              .3
i9   3.5  2.28 .5   0    .9   .1   .5             .1
i9   4.5  2.28 .5   0    1.2  .1   0              .2
i9   5    2.28 .5   0    .2   .1   1              .3
i9   9    2.28 .7   0    .1   .1   .5             .03
```

toot09.sco

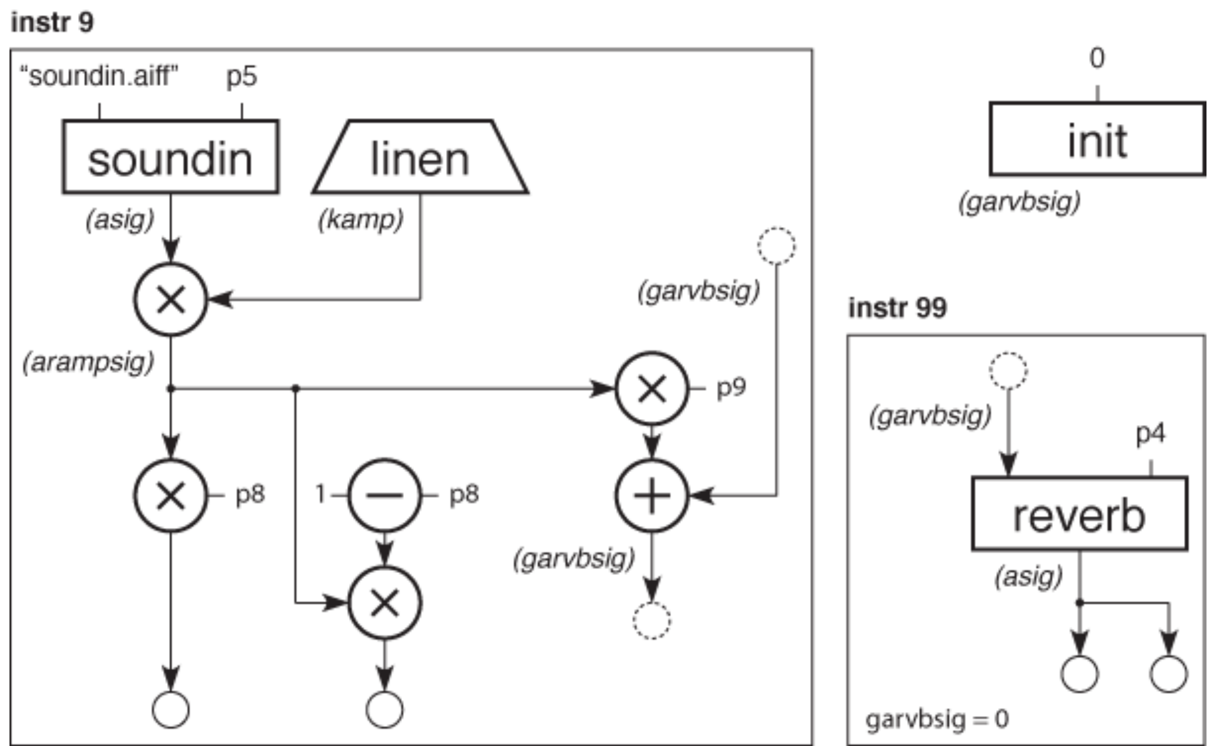


Figure 9 Block diagram of instr 9.

Toot 10: Filtered Noise

The following instrument uses the Csound **rand** unit to produce noise, and a **reson** unit to filter it. The bandwidth of **reson** will be set at i-time, but its center frequency will be swept via a **line** unit through a wide range of frequencies during each note. We add reverb as in Toot 9.

```
          instr 10
iattack   =         .01
irelease  =         .2
iwhite    =         22050
idur      =         p3
iamp      =         p4
iswpstart =         p5
isweepend =         p6
ibndwidth =         p7
ibalance  =         p8                  ; 1 = left, .5 = center, 0 = right
irvbgain  =         p9
kamp      linen     iamp, iattack, idur, irelease
ksweep    line      iswpstart, idur, isweepend
asig      rand      iwhite
afilt     reson     asig, ksweep, ibndwidth
arampsig  =         kamp * afilt
          outs      arampsig * ibalance, arampsig * (1 - ibalance)
garvbsig  =         garvbsig + arampsig * p9
          endin

          instr     100
irvbtime  =         p4
asig      reverb    garvbsig,  irvbtime
          outs      asig, asig
garvbsig  =         0
          endin


;ins strt dur  rvbtime
i100 0    15   1.1
i100 15   10   5

;ins strt dur  amp   stsw   ndsw   bdw  bal(0-1)  rvsnd
i10  0    2    .01   5000   500    20   .15        .1
i10  3    1    .01   1500   5000   30   .95        .1
i10  5    2    .01   850    1100   40   .45        .1
i10  8    2    .01   1100   8000   50   .05        .1
i10  8    .5   .01   5000   1000   30   .35        .2
i10  9    .5   .01   1000   8000   40   .75        .1
```

```
i10  11   .5   .01   500    2100   50   .14         .2
i10  12   .5   .01   2100   1220   75   .96         .1
i10  13   .5   .01   1700   3500   100  .45         .2
i10  15   5    .005  8000   800    60   .85         .1
```

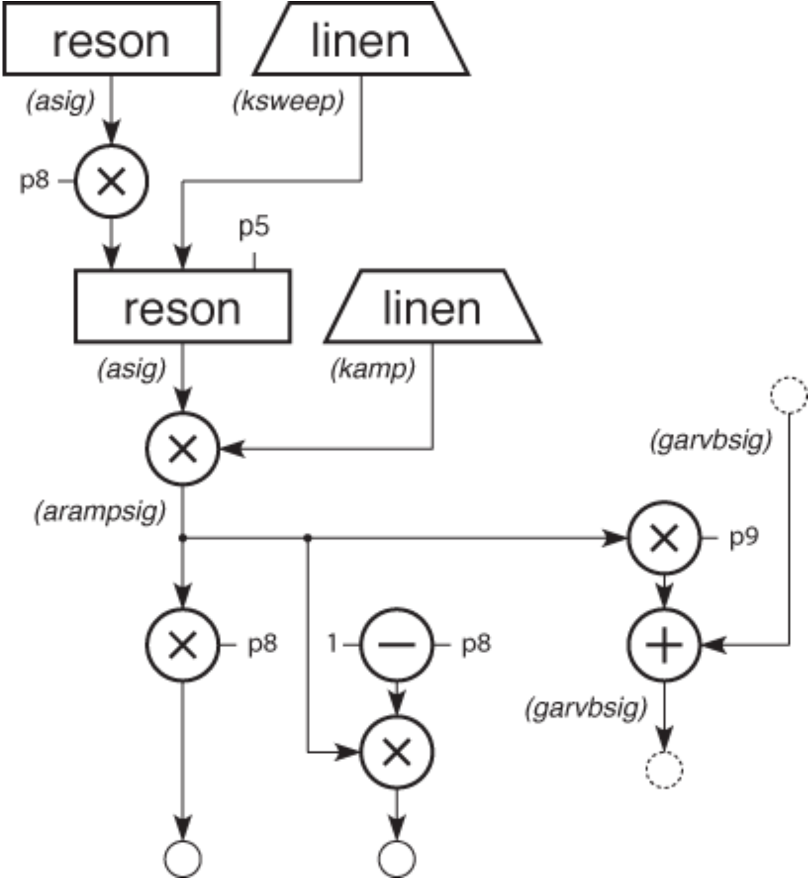


Figure 10 Block diagram of instr 10.

Toot 11: Carry, Tempo & Sort

We now use a plucked string instrument to explore some of Csound's score preprocessing capabilities. Since the focus here is on the score, the instrument is presented without explanation.

```
        instr 11
asig1   pluck   ampdb(p4)/2, p5, p5, 0, 1
asig2   pluck   ampdb(p4)/2, p5*1.003, p5*1.003, 0, 1
        out     asig1 + asig2
        endin
```

The score can be divided into time-ordered sections by the **s** statement**.** Prior to performance, each section is processed by three routines: **Carry**, **Tempo**, and **Sort**. The score toot11.sco has multiple sections containing each of the examples below, in both of the forms listed.

Carry

The carry feature allows a dot (".") in a p-field to indicate that the value is the same as above, provided the instrument is the same. Thus the following two examples are identical:

```
;ins start dur  amp  freq   |    ; ins start dur   amp  freq
 i11    0    1   90   200   |      i11    0    1    90   200
 i11    1    .    .   300   |      i11    1    1    90   300
 i11    2    .    .   400   |      i11    2    1    90   400
```

A special form of the carry feature applies to p2 only. A "+" in p2 will be given the value of p2+p3 from the previous **i** statement. The "+" can also be carried with a dot:

```
;ins start dur  amp  freq   |   ;  ins start  dur  amp  freq
 i11    0    1   90   200   |      i11    0    1    90   200
 i.     +    .    .   300   |      i11    1    1    90   300
 i.     .    .    .   500   |      i11    2    1    90   500
```

The carrying dot may be omitted when there are no more explicit pfields on that line:

Ramping

A variant of the carry feature is ramping, which substitutes a sequence of linearly interpolated values for a ramp symbol (“<”) spanning any two values of a pfield. Ramps work only on consecutive calls to the same instrument, and they cannot be applied to the first three p-fields.

```
;ins start dur  amp  freq   |    ; ins start  dur  amp  freq
 i11    0    1   90   200   |      i11     0    1   90   200
 i .    +    .   <    <     |      i11     1    1   85   300
 i .    .    .   <    400   |      i11     2    1   80   400
 i .    .    .   <    <     |      i11     3    1   75   300
 i .    .    4   70   200   |      i11     4    4   70   200
```

Tempo

The unit of time in a Csound score is the beat - normally one beat per second. This can be modified by a tempo statement which enables the score to be arbitrarily time-warped. Beats are converted to their equivalent in seconds during score pre-processing of each Section. In the absence of a Tempo statement in any Section, the following tempo statement is inserted:

```
t 0 60
```

It means that at beat 0 the tempo of the Csound beat is 60 (1 beat per second). To hear the Section at twice the speed, we have two options: 1) cut all p2 and p3 in half and adjust the start times, or 2) insert the statement t 0 120 within the Section.

The tempo statement can also be used to move between different tempi during the score, thus enabling ritardandi and accelerandi. Changes are linear by beat size. The following statement will cause the score to begin at tempo 120, slow to tempo 80 by beat 4, then accelerate to 220 by beat 7:

```
t 0 120 4 80 7 220
```

The following will produce identical sound files:

```
                                |  t  0  120       ; Double-time via Tempo
;ins  start  dur  amp  freq    |  ; ins start  dur  amp  freq
 i11     0    .5   90   200    |    i11     0    1   90   200
 i .     +    .    <    <      |    i .     +    .   <    <
 i .     .    .    <    400    |    i .     .    .   <    400
 i .     .    .    <    <      |    i .     .    .   <    <
 i .     .    2    70   200    |    i .     .    4   70   200
```

The following includes an accelerando and ritard. It should be noted, however, that the ramping feature is applied *after* time-warping, and is thus proportional to elapsed chronological time. While this is perfect for amplitude ramps, frequency ramps will not result in harmonically related pitches during tempo changes. The frequencies needed here are thus made explicit.

```
t 0 60 4 400 8 60  ; Time-warping via Tempo

;ins start dur  amp  freq
 i11  0    1    70   200
 i .  +    .    <    500
 i .  .    .    90   800
 i .  .    .    <    500
 i .  .    .    70   200
 i .  .    .    90   1000
 i .  .    .    <    600
 i .  .    .    70   200
 i .  .    8    90   100
```

Three additional score features are extremely useful in Csound. The **s** statement was used above to divide a score into Sections for individual pre-processing. Since each s statement establishes a new relative time of 0, and all actions within a section are relative to that, it is convenient to develop the score one section at a time, then link the sections into a whole later.

Suppose we wish to combine the six above examples (call them toot11a - toot11f) into one score. One way is to start with toot11a.sco, calculate its total duration and add that value to every starting time of toot11b.sco, then add the composite duration to the start times of toot11c.sco, etc. Alternatively, we could insert an **s** statement between each of the sections and run the entire score. The file toot11.sco, which contains a sequence of all of the above score examples, did just that.

The **f0** statement, which creates an "action time" with no associated action, is useful in

extending the duration of a section. Two seconds of silence are added to the first two sections below.

Sort

During preprocessing of a score section, all action-time statements are sorted into chronological order by p2 value. This means that notes can be entered in any order, that you can merge files, or work on instruments as temporarily separate sections, then have them sorted automatically when you run Csound on the file.

The file below contains excerpts from this section of the rehearsal chapter and from instr6 of the tutorial, and combines them as follows:

```
;   ins start dur  amp  freq           ; toot11h.sco
    i11  0    1    70   100            ; Score Sorting
    i .  +    .    <    <
    i .  .    .    <    <
    i .  .    .    90   800
    i .  .    .    <    <
    i .  .    .    <    <
    i .  .    .    70   100
    i .  .    .    90   1000
    i .  .    .    <    <
    i .  .    .    <    <
    i .  .    .    <    <
    i .  .    .    70   <
    i .  .    8    90   50

   f1 0 2048 10 1                            ; Sine
   f2 0 2048 10 1 .5 .3 .25 .2 .167 .14 .125 .111      ; Sawtooth
   f3 0 2048 10 1 0  .3  0   .2  0 .14  0   .111       ; Square
   f4 0 2048 10 1 1 1 1 .7 .5 .3 .1                    ; Pulse

; ins  strt  dur   amp   frq  atk rel vibr vibdpth vibdel    waveform
   i6     0    2    86   9.00 .03  .1   6    5         .4      1
   i6     2    2    86   9.02 .03  .1   6    5         .4      2
   i6     4    2    86   9.04 .03  .1   6    5         .4      3
   i6     6    4    86   9.05 .05  .1   6    5         .4      4
```

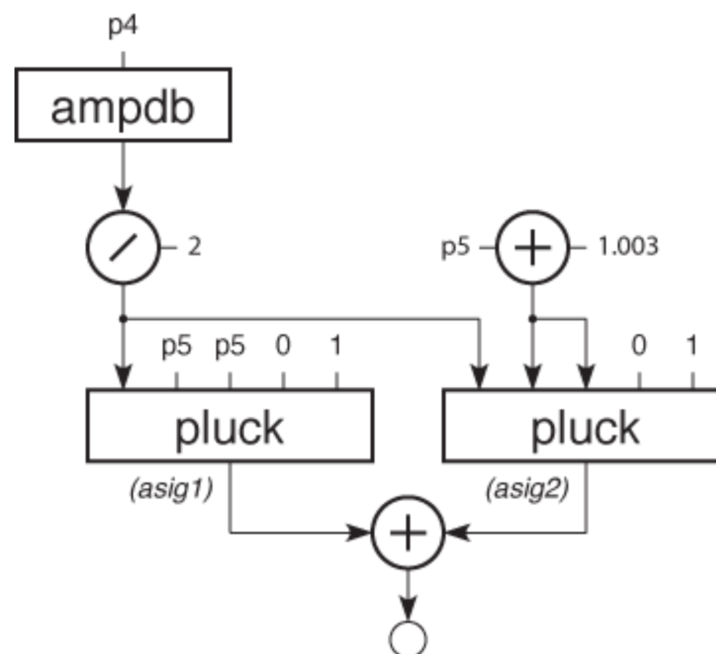


Figure 11 Block diagram of instr 11.

Toot 12: Tables & Labels

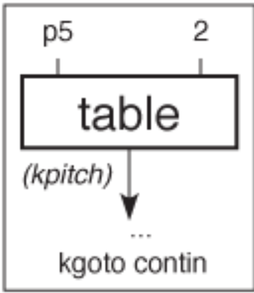
This is by far our most complex instrument. In it we have designed the ability to store pitches in a table, and then index them in three different ways: 1) directly, 2) via an lfo, and 3) randomly. As a means of switching between these three methods, we will use Csound's *program control* statements and *logical* and *conditional* operations.

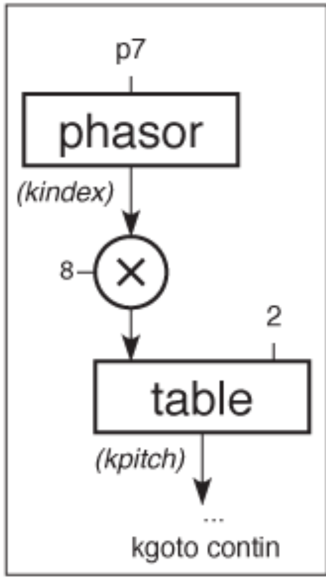
```
          instr 12
iseed     =         p8
iamp      =         ampdb(p4)
kdirect   =         p5
imeth     =         p6
ilforate  =         p7                              ; rate for lfo and random index
```

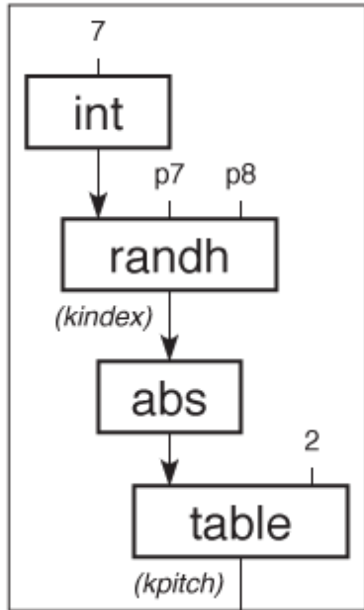
```
          itab      =         2
          itabsize  =         8

if (imeth == 1)     igoto     direct
if (imeth == 2)     kgoto     lfo
if (imeth == 3)     kgoto     random

direct:   kpitch    table     kdirect, itab                 ; index f2 via p5
                    kgoto     contin

lfo:      kindex    phasor    ilforate
          kpitch    table     kindex * itabsize, itab
                    kgoto     contin

random:   kindex    randh     int(7), ilforate, iseed
          kpitch    table     abs(kindex), itab

contin:   kamp      linseg    0, p3 * .1, iamp, p3 * .9, 0  ; amp envelope
          asig      oscil     kamp, cpspch(kpitch), 1       ; audio oscillator
                    out       asig
                    endin


f1   0    4096 10 1                                         ; Sine
f2   0    8    -2 8.00 8.02 8.04 8.05 8.07 8.09 8.11 9.00   ; cpspch C major scale

; method 1 - direct index of table values
; ins     strt dur  amp  index     method    lforate   rndseed
  i12     0    .5   86   7         1         0         0
  i12     .5   .5   86   6         1         0
  i12     1    .5   86   5         1         0
  i12     1.5  .5   86   4         1         0
  i12     2    .5   86   3         1         0
  i12     2.5  .5   86   2         1         0
  i12     3    .5   86   1         1         0
  i12     3.5  .5   86   0         1         0
  i12     4    .5   86   0         1         0
  i12     4.5  .5   86   2         1         0
  i12     5    .5   86   4         1         0
  i12     5.5  2.5  86   7         1         0
  f0      10
s
; method 2 - lfo index of table values
; ins     strt dur  amp  index     method    lforate   rndseed
  i12     0    2    86   0         2         1         0
  i12     3    2    86   0         2         2
```

```
  i12     6    2    86   0         2         4
  i12     9    2    86   0         2         8
  i12     12   2    86   0         2         16
  f0      16
s
; method 3 - random index of table values
; ins     strt dur  amp  index     method    lforate   rndseed
  i12     0    2    86   0         3         2         .1
  i12     3    2    86   0         3         3         .2
  i12     6    2    86   0         3         4         .3
  i12     9    2    86   0         3         7         .4
  i12     12   2    86   0         3         11        .5
  i12     15   2    86   0         3         18        .6
  i12     18   2    86   0         3         29        .7
  i12     21   2    86   0         3         47        .8
  i12     24   2    86   0         3         76        .9
  i12     27   2    86   0         3         123       .9
  i12     30   5    86   0         3         199       .
```

```
if (imeth == 1)   igoto   direct
if (imeth == 2)   kgoto   lfo
if (imeth == 3)   kgoto   random
```







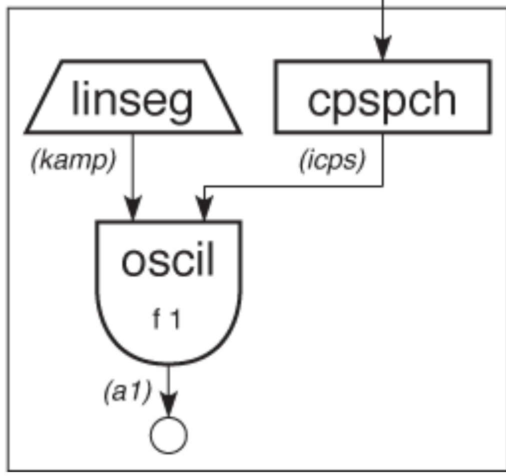


Figure 12 Block diagram of instr 12.

Toot 13: Spectral Fusion

For our final instrument, we will employ three unique synthesis methods: Physical Modeling, Formant-Wave Synthesis, and Non-linear Distortion. Three of Csound's more powerful unit generators - **pluck**, **fof**, and **foscil**, make this complex task a fairly simple one. The Reference Manual describes these as follows:

```
ar pluck kamp, kcps, icps, ifn, imeth\\
             [, iparm1, iparm2]
```

pluck simulates the sound of naturally decaying plucked strings by filling a cyclic decay buffer with noise and then smoothing it over time according to one of several methods. The unit is based on the Karplus-Strong algorithm.

```
ar fof xamp, xfund, xform, koct, kband, kris,\\
             kdur, kdec, iolaps, ifna, ifnb, itotdur\\
             [, iphs[, ifmode]]
```

fof simulates the sound of the male voice by producing a set of harmonically related partials (a formant region) whose spectral envelope can be controlled over time. It is a special form of granular synthesis, based on the CHANT program from IRCAM by Xavier Rodet et al.

```
ar foscil xamp, kcps, kcar, kmod, kndx, ifn\\
             [, iphs]
```

foscil is a composite unit which banks two oscillators in a simple FM configuration, wherein the audio-rate output of one (the “modulator”) is used to modulate the frequency input of another (the “carrier.”)

The plan for our instrument is to have the plucked string attack dissolve into an FM sustain

which transforms into a vocal release. The orchestra and score are as follows:

```
          instr 13
iamp      =         ampdb(p4) / 2       ;amp scaled for two sources
ipluckamp =         p6                  ;p6: % of total amplitude, 1=dB amp as in p4
ipluckdur =         p7*p3               ;p7: % of total duration, 1=entire note duration
ipluckoff =         p3 - ipluckdur

ifmamp    =         p8                  ;p8: % of total amplitude, 1=dB amp as in p4
ifmrise   =         p9*p3               ;p9: % of total duration, 1=entire note duration
ifmdec    =         p10*p3              ;p10: % of total duration
ifmoff    =         p3 - (ifmrise + ifmdec)
index     =         p11
ivibdepth =         p12
ivibrate  =         p13
ifrmntamp =         p14                 ;p14: % of total amplitude, 1=dB amp as in p4
ifrmntris =         p15*p3              ;p15: % of total duration, 1=entire note duration
ifrmntdec =         p3 - ifrmntris

kpluck    linseg    ipluckamp, ipluckdur, 0, ipluckoff, 0
apluck1   pluck     iamp, p5, p5, 0, 1
apluck2   pluck     iamp, p5*1.003, p5*1.003, 0, 1
apluck    =         kpluck * (apluck1+apluck2)

kfm       linseg    0, ifmrise, ifmamp, ifmdec, 0, ifmoff, 0
kndx      =         kfm * index
afm1      foscil    iamp, p5, 1, 2, kndx, 1
afm2      foscil    iamp, p5*1.003, 1.003, 2.003, kndx, 1
afm       =         kfm * (afm1+afm2)

kformant linseg     0, ifrmntris, ifrmntamp, ifrmntdec, 0
kvib      oscil     ivibdepth, ivibrate, 1
afrmt1    fof       iamp, p5+kvib, 650, 0, 40, .003,.017,.007,4,1,2,p3
afrmt2    fof       iamp, (p5*1.001)+kvib*.009, 650, 0, 40, .003,.017,.007,10,1,2,p3
aformant  =         kformant * (afrmt1+afrmt2)

          out       apluck + afm + aformant
          endin


f1   0    8192 10   1                                                                ; Sine
f2   0    2048 19   0.5    1  270 1                                                  ; Sine quadra

;  pluckamp = p6         -    % of total amplitude, 1=dB amp as specified in p4
;  pluckdur = p7*p3      -    % of total duration, 1=entire duration of note
```

```
;  fmamp = p8            -    % of total amplitude, 1=dB amp as specified in p4
;  fmrise = p9*p3        -    % of total duration, 1=entire duration of note
;  fmdec = p10*p3        -    % of total duration
;  index = p11           -    number of significant sidebands: p11 + 2
;  vibdepth = p12
;  vibrate = p13
;  formantamp = p14      -    % of total amplitude, 1=dB amp as specified in p4
;  formantrise = p15*p3  -    % of total duration, 1=entire duration of note

 f0 01
 f0 02
 f0 03
 f0 04
 f0 06
 f0 07
 f0 08
 f0 09
 f0 10
 f0 11
 f0 12
 f0 14
 f0 15
 f0 16
 f0 17
 f0 18
 f0 19
 f0 20
 f0 21
 f0 22
 f0 23
 f0 24
 f0 25

;ins  st  dur  amp  frq  plkmp  plkdr  fmmp   fmrs    fmdc  indx  vbdp      vbrt  frmp  fris
i13   0    5   80   200   .8     .3      .7   .2      .35     8     1       5       3   .5
i13   5    8   80   100   .      .4      .7   .35     .35     7     1       6       3   .7
i13  13   13   80    50   .      .3      .7   .2      .4      6     1       4       3   .6
```

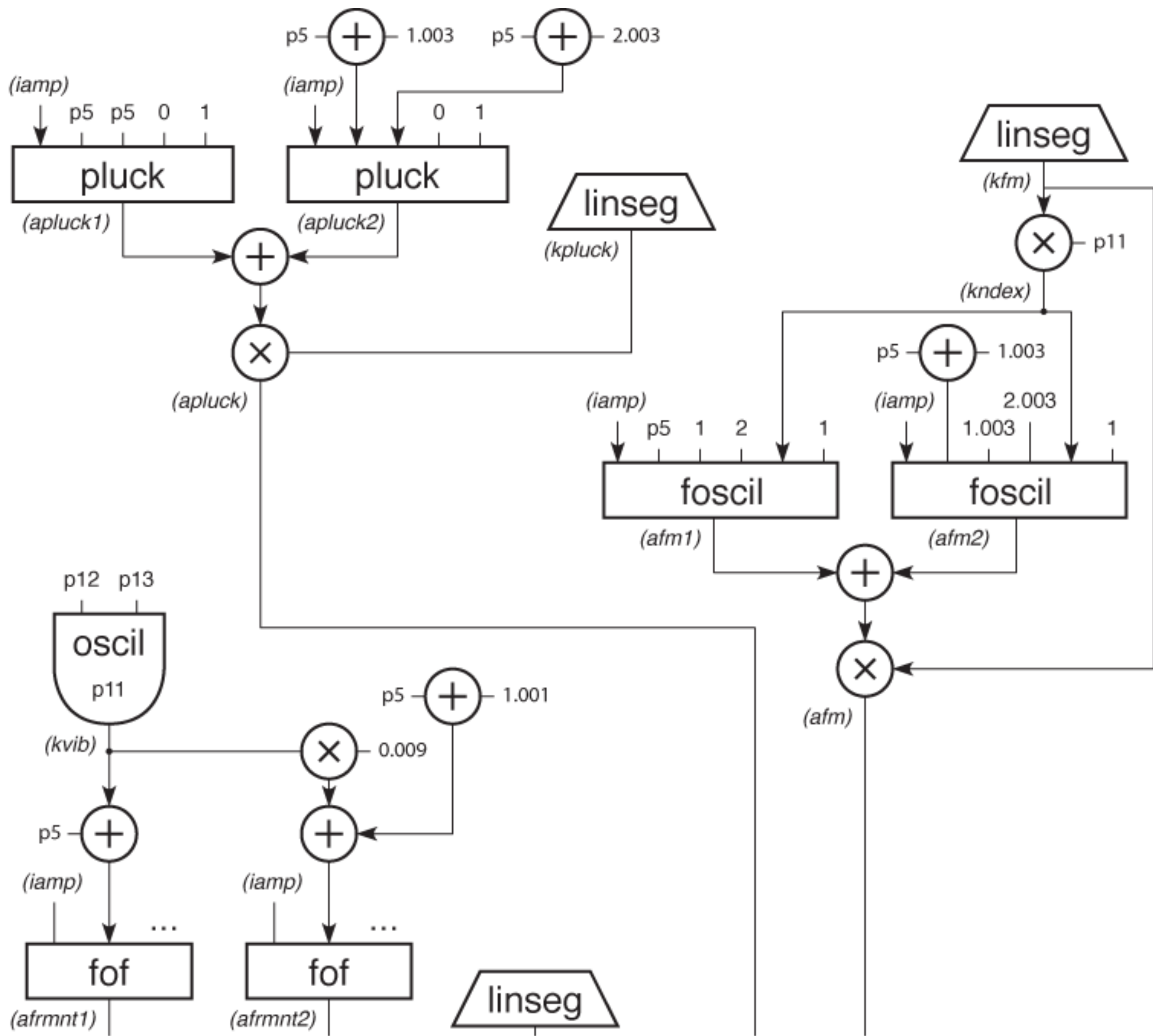

p5
1.003
p5
2.003
(iamp)
p5 p5 0 1
pluck
(iamp)
0 1
pluck
linseg
(kfm)
p11
(apluck1)
(apluck2)
linseg
(kpluck)
(kndex)
p5
1.003
(apluck)
(iamp)
p5 1 2 1
(iamp)
2.003
1.003
1
foscil
foscil
(afm1)
(afm2)
p12 p13
oscil
p11
p5
1.001
(afm)
(kvib)
0.009
p5
(iamp)
(iamp)
...
...
fof
fof
linseg
(afrmnt1)
(afrmnt2)

Figure 13 Block diagram of instr 13.

When Things Sound Wrong

When you design your own Csound instruments you may occasionally be surprised by the results. There will be times when you've computed a file for hours and your playback is just silence, while at other times you may get error messages which prevent the score from running, or you may hang the computer and nothing happens at all.

In general, Csound has a comprehensive error-checking facility that reports to your console at various stages of your run: at score sorting, orchestra translation, initializing each call of every instrument, and during performance. However, if your error was syntactically permissable, or it generated only a warning message, Csound could faithfully give you results you don't expect. Here is a list of the things you might check in your score and orchestra files:

- You typed the letter “l” instead of the number “1.”
- You forgot to precede your comment with a semi-colon.
- You forgot an opcode or a required parameter.
- Your amplitudes are not loud enough, or they are too loud.
- Your frequencies are not in the audio range - 20Hz to 20kHz.
- You placed the value of one parameter in the p-field of another.
- You left out some crucial information like a function definition.
- You didn't meet the GEN specifications.

Suggestions for Further Study

Csound is such a powerful tool that we have touched on only a few of its many features and uses. You are encouraged to take apart the instruments in the tutorials, rebuild them, modify them, and

integrate the features of one into the design of another. To understand their capabilities you should compose short etudes with each. You may be surprised to find yourself merging these little studies into the fabric of your first Csound compositions.

There are many sources of information on Csound and software synthesis. The ultimate sourcebook for Csound is *The Csound Book: Perspectives in Software Synthesis, Sound Design, Signal Processing, and Programming*, edited by Richard Boulanger, and published by MIT Press.

Nothing will increase your understanding more than actually making music with Csound. The best way to discover the full capability of these tools is to create your own music with them. As you negotiate the new and uncharted terrain you will make many discoveries. It is my hope that through Csound you discover as much about music as I have, and that this experience brings you great personal satisfaction and joy.

Richard Boulanger
Boston, Massachusetts USA
March, 1991