Csound

A Manual for the Audio Processing System

and

Supporting Programs

with

Tutorials

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0. PREFACE

Realizing music by digital computer involves synthesizing audio signals with discrete points or *samples* that are representative of continuous waveforms. There are several ways of doing this, each affording a different manner of control. *Direct synthesis* generates waveforms by sampling a stored *function* representing a single cycle; *additive synthesis* generates the many partials of a complex tone, each with its own loudness envelope; *subtractive synthesis* begins with a complex tone and filters it. *Non-linear synthesis* uses frequency modulation and waveshaping to give simple signals complex characteristics, while *sampling* and storage of natural sound allows it to be used at will.

Since comprehensive moment-by-moment specification of sound can be tedious, control is gained in two ways: 1) from the *instruments* in an *orchestra*, and 2) from the *events* within a *score*. An orchestra is really a computer program that can produce sound, while a score is a body of data which that program can react to. Whether a rise-time characteristic is a fixed constant in an instrument, or a variable of each note in the score, depends on how the user wants to control it.

The instruments in a **Csound** orchestra are defined in a simple syntax that invokes complex audio processing routines. A score passed to this orchestra contains numerically coded pitch and control information, in *standard numeric score* format. Although many users are content with this format, higher level score processing languages are often convenient.

The programs making up the **Csound** system have a long history of development, beginning with the *Music 4* program written at Bell Telephone Laboratories in the early 1960's by Max Mathews. That initiated the stored table concept and much of the terminology that has since enabled computer music researchers to communicate. Valuable additions were made at Princeton by the late Godfrey Winham in *Music 4B*; my own *Music 360* (1968) was very indebted to his work. With *Music 11* (1973) I took a different tack: the two distinct networks of *control* and *audio* signal processing stemmed from my intensive involvement in the preceding years in hardware synthesizer concepts and design. This division has been retained in **Csound**.

Because it is written entirely in **C, Csound** is easily installed on any machine running Unix or C. At MIT it runs on VAX/DECstations under Ultrix 4.2, on SUNs under OS 4.1, SGI's under 4.1, and on the Macintosh under ThinkC 4.0. With this single language for audio signal processing, users move easily from machine to machine.

The **1991** version included many new features. I am indebted to others for the contribution of the phase vocoder and FOF synthesis modules. That release also charted a new direction with the addition of a *spectral data type*, holding much promise for future development. The **1992** release is even more significant for its addition of MIDI converter and control units, enabling **Csound** to be run from MIDI score-files and from external MIDI keyboards. Since the newest RISC processors bring to computer music an order of magnitude more speed than did those on which it was born, researchers and composers now have access to workstations on which *realtime software synthesis with sensing and control* is now a reality. This is perhaps the single most important development for people working in the field. This new Csound is designed to take maximum advantage of realtime audio processing, and to encourage interactive experiments in this exciting new domain.

B.V.

1. A BEGINNING TUTORIAL

Introduction

The purpose of this section is to expose the reader to the fundamentals of designing and using computer music instruments in **Csound**. Only a small portion of the language will be covered here, sufficient to implement some simple instrument examples. The sections in this primary text are arranged as a *Reference* manual (not a tutorial), since that is the form the user will eventually find most helpful when inventing instruments. Once the basic concepts are grasped from this beginning tutorial, the reader might let himself into the remainder of the text by locating the information presented here in the Reference entries that follow. More comprehensive tutorials are supplied as Appendices.

The Orchestra File

Csound runs from two basic files: an *orchestra* file and a *score* file. The orchestra file is a set of *instruments* that tell the computer how to synthesize sound; the score file tells the computer when. An instrument is a collection of modular statements which either *generate* or *modify* a signal; signals are represented by *symbols*, which can be "patched" from one module to another. For example, the following two statements will generate a 440 Hz sine tone and send it to an output channel:

```
asig    oscil   10000, 440, 1
        out     asig
```

The first line sets up an oscillator whose controlling inputs are an amplitude of 10000, a frequency of 440 Hz, and a waveform number, and whose output is the audio signal *asig*. The second line takes the signal *asig* and sends it to an (implicit) output channel. The two may be encased in another pair of statements that identify the instrument as a whole:

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

In general, an orchestra statement in **Csound** consists of an action symbol followed by a set of input variables and preceded by a result symbol. Its *action* is to process the inputs and deposit the result where told. The meaning of the input variables depends on the action requested. The 10000 above is interpreted as an amplitude value because it occupies the first input slot of an oscil unit; 440 signifies a frequency in Hertz because that is how an oscil unit interprets its second input argument; the waveform number is taken to point indirectly to a stored function table, and before we invoke this instrument in a score we must fill function table #1 with some waveform.

The output of **Csound** computation is not a real audio signal, but a stream of numbers which describe such a signal. When written onto a sound file these can later be converted to sound by an independent program; for now, we will think of variables such as *asig* as tangible audio signals.

Let us now add some extra features to this instrument. First, we will allow the pitch of the tone to be defined as a *parameter* in the score. Score parameters can be represented by orchestra variables which take on their different values on successive notes. These variables are named sequentially: p1, p2, p3, ... The first three have a fixed meaning (see the Score File), while the remainder are assignable by the user. Those of significance here are:

p3 - duration of the current note (always in seconds).
p5 - pitch of the current note (in units agreed upon by score and orchestra).

Thus in

```
asig    oscil   10000, p5, 1
```

the oscillator will take its pitch (presumably in cps) from score parameter 5.

If the score had forwarded pitch values in units other than cycles-per-second (Hertz), then these must first be converted. One convenient score encoding, for instance, combines *pitch class* representation (00 for C, 01 for C#, 02 for D, ... 11 for B) with *octave* representation (8. for middle C, 9. for the C above, etc.) to give pitch values such as 8.00, 9.03, 7.11. The expression

```
cpspch(8.09)
```

will convert the pitch A (above middle C) to its cps equivalent (440 Hz). Likewise, the expression

```
cpspch(p5)
```

will first read a value from p5, then convert it from octave.pitch-class units to cps. This expression could be imbedded in our orchestra statement as

```
asig    oscil   10000, cpspch(p5), 1
```

to give the score-controlled frequency we sought.

Next, suppose we want to shape the amplitude of our tone with a linear rise from 0 to 10000. This can be done with a new orchestra statement

```
amp     line    0, p3, 10000
```

Here, *amp* will take on values that move from 0 to 10000 over time p3 (the duration of the note in seconds). The instrument will then become

```
        instr   1
amp     line    0, p3, 10000
asig    oscil   amp, cpspch(p5), 1
        out     asig
        endin
```

The signal *amp* is not something we would expect to listen to directly. It is really a variable whose purpose is to control the amplitude of the audio oscillator. Although audio output requires fine resolution in time for good fidelity, a controlling signal often does not need that much resolution. We could use another kind of signal for this amplitude control

```
kamp    line    0, p3, 10000
```

in which the result is a new kind of signal. Signal names up to this point have always begun with the letter **a** (signifying an *audio* signal); this one begins with **k** (for *control*). Control signals are identical to audio signals, differing only in their resolution in time. A control signal changes its value less often than an audio signal, and is thus faster to generate. Using one of these, our instrument would then become

```
         instr   1
kamp     line    0, p3, 10000
asig     oscil   kamp, cpspch(p5), 1
         out     asig
         endin
```

This would likely be indistinguishable in sound from the first version, but would run a little faster. In general, instruments take constants and parameter values, and use calculations and signal processing to move first towards the generation of control signals, then finally audio signals. Remembering this flow will help you write efficient instruments with faster execution times.

We are now ready to create our first orchestra file. Type in the following orchestra using the system editor, and name it "intro.orc".

```
         sr = 20000                   ; audio sampling rate is 20 kHz
         kr = 500                     ; control rate is 500 Hz
         ksmps = 40                   ; number of samples in a control period (sr/kr)
         nchnls = 1                   ; number of channels of audio output

         instr   1
kctrl    line    0, p3, 10000         ; amplitude envelope
asig     oscil   kctrl, cpspch(p5), 1 ; audio oscillator
         out     asig                 ; send signal to channel 1
         endin
```

It is seen that comments may follow a semi-colon, and extend to the end of a line. There can also be blank lines, or lines with just a comment. Once you have saved your orchestra file on disk, we can next consider the score file that will drive it.

The Score File

The purpose of the score is to tell the instruments when to play and with what parameter values. The score has a different syntax from that of the orchestra, but similarly permits one statement per line and comments after a semicolon. The first character of a score statement is an **opcode**, determining an action request; the remaining data consists of numeric parameter fields (pfields) to be used by that action.

Suppose we want a sine-tone generator to play a pentatonic scale starting at C-sharp above middle-C, with notes of 1/2 second duration. We would create the following score:

```
;  a sine wave function table
f1 0 256 10 1
;  a pentatonic scale
i1   0  .5     0.     8.01
i1  .5  .      .      8.03
i1 1.0  .      .      8.06
i1 1.5  .      .      8.08
i1 2.0  .      .      8.10
e
```

The first statement creates a stored sine table. The protocol for generating wave tables is simple but powerful. Lines with opcode **f** interpret their parameter fields as follows:

p1 - function table *number* being created
p2 - *creation time*, or time at which the table becomes readable
p3 - table *size* (number of points), which must be a power of two or one greater
p4 - *generating subroutine*, chosen from a prescribed list.

Here the value 10 in p4 indicates a request for subroutine **GEN10** to fill the table. **GEN10** mixes harmonic sinusoids in phase, with relative strengths of consecutive partials given by the succeeding parameter fields. Our score requests just a single sinusoid. An alternative statement:

```
f1 0 256 10 1 0 3
```

would produce one cycle of a waveform with a third harmonic three times as strong as the first.

The **i** statements, or note statements, will invoke the p1 instrument at time p2, then turn it off after p3 seconds; it will pass all of its p-fields to that instrument. Individual score parameters are separated by any number of spaces or tabs; neat formatting of parameters in columns is nice but unnecessary. The dots in p-fields 3 and 4 of the last four notes invoke a *carry feature*, in which values are simply copied from the immediately preceding note *of the same instrument*. A score normally ends with an **e** statement.

The unit of time in a **Csound** score is the beat. .In the absence of a *Tempo* statement, one beat takes one second. To double the speed of the pentatonic scale in the above score, we could either modify p2 and p3 for all the notes in the score, or simply insert the line

```
t 0 120
```

to specify a tempo of 120 beats per minute from beat 0.

Two more points should be noted. First, neither the *f*-statements nor the *i*-statements need be typed in time order; **Csound** will sort the score automatically before use. Second, it is permissible to play more than one note at a time with a single instrument. To play the same notes as a three-second pentatonic chord we would create the following:

```
;   a sine wave function
f1 0 256 10 1
;   five notes at once
i1  0   3       0       8.01
i1  0   .       .       8.03
i1  0   .       .       8.06
i1  0   .       .       8.08
i1  0   .       .       8.10
e
```

Now go into the editor once more and create your own score file. Name it "intro.sco". Tne next section will describe how to invoke a **Csound** orchestra to perform a **Csound** score.

The CSOUND Command

To request your orchestra to perform your score, type the command

csound intro.orc intro.sco

The resulting performance will take place in three phases:

1) sort the score file into chronological order. If score syntax errors are encountered they will be reported on your console.

2) translate and load your orchestra. The console will signal the start of translating each *instr* block, and will report any errors. If the error messages are not immediately meaningful, translate again with the *verbose* flag turned on:

csound -v intro.orc intro.sco

3) fill the wave tables and perform the score. Information about this performance will be displayed throughout in messages resembling

B 4.000 .. 6.000 T 3.000 TT 3.000 M 7929. 7929.

A message of this form will appear for every *event* in your score. An event is defined as any change of state (as when a new note begins or an old one ends). The first two numbers refer to beats in your original score, and they delimit the current segment of sound synthesis between successive events (e.g. from beat 4 to beat 6). The second beat value is next restated in real seconds of time, and reflects the *tempo* of the score. That is followed by the Total Time elapsed for all sections of the score so far. The last values on the line show the maximum amplitude of the audio signal, measured over just this segment of time, and reported separately for each channel.

Console messages are printed to assist you in following the orchestra's handling of your score. You should aim at becoming an intelligent reader of your console reports. When you begin working with longer scores and your instruments no longer cause surprises, the above detail may be excessive. You can elect to receive abbreviated messages using the *-m* option of the **Csound** command.

When your performance goes to completion, it will have created a sound file named *test* in your soundfile directory. You can now listen to your sound file by typing

play test

If your machine is fast enough, and your Csound module includes user access to the audio output device, you can hear your sound as it is being synthesized by using a command like

csound -o devaudio intro.orc intro.sco

More about the Orchestra

Suppose we next wished to introduce a small vibrato, whose rate is 1/50 the frequency of the note (i.e. A440 is to have a vibrato rate of 8.8 Hz.). To do this we will generate a control signal using a second oscillator, then add this signal to the basic frequency derived from p5. This might result in the instrument

```
        instr   1
kamp    line    0, p3, 10000
kvib    oscil   2.75, cpspch(p5)/50, 1
a1      oscil   kamp, cpspch(p5)+kvib, 1
        out     a1
        endin
```

Here there are two control signals, one controlling the amplitude and the other modifying the basic pitch of the audio oscillator. For small vibratos, this instrument is quite practical; however it does contain a misconception worth noting. This scheme has added a sine wave deviation to the cps value of an audio oscillator. The value 2.75 determines the *width* of vibrato in cps, and will cause an A440 to be modified about one-tenth of one semitone in each direction (1/160 of the frequency in cps). In reality, a cps deviation produces a different musical interval above than it does below. To see this, consider an exaggerated deviation of 220 cps, which would extend a perfect 5th above A440 but a whole octave below. To be more correct, we should first convert p5 into a *true decimal octave* (not cps), so that an *interval* deviation above is equivalent to that below. In general, pitch modification is best done in true octave units rather than pitch-class or cps units, and there exists a group of pitch converters to make this task easier. The modified instrument would be

```
        instr   1
ioct    =       octpch(p5)
kamp    line    0, p3, 10000
kvib    oscil   1/120, cpspch(p5)/50, 1
asig    oscil   kamp, cpsoct(ioct+kvib), 1
        out     asig
        endin
```

This instrument is seen to use a third type of orchestra variable, an *i*-variable. The variable *ioct* receives its value at an *initialization* pass through the instrument, and does not change during the lifespan of this note. There may be many such *init time* calculations in an instrument. As each note in a score is encountered, the event space is allocated and the instrument is initialized by a special pre-performance pass. *i-variables* receive their values at this time, and any other expressions involving just constants and *i*-variables are evaluated. At this time also, modules such as **line** will set up their target values (such as beginning and end points of the line), and units such as **oscil** will do phase setup and other bookkeeping in preparation for performance. A full description of init-time and performance-time activities, however, must be deferred to a general consideration of the orchestra syntax.

2. SYNTAX OF THE ORCHESTRA

An orchestra statement in **Csound** has the format:

label: result opcode argument1, argument2, ... ; comments

The label is optional and identifies the basic statement that follows as the potential target of a go-to operation (see Program Control Statements). A label has no effect on the statement per se.

Comments are optional and are for the purpose of letting the user document his orchestra code. Comments always begin with a semicolon (;) and extend to the end of the line.

The remainder (result, opcode, and arguments) form the *basic statement*. This also is optional, i.e. a line may have only a label or comment or be entirely blank. If present, the basic statement must be complete on one line. The opcode determines the operation to be performed; it usually takes some number of input values (arguments); and it usually has a result field variable to which it sends output values at some fixed rate. There are four possible rates:

1) once only, at orchestra setup time (effectively a permanent assignment);
2) once at the beginning of each note (at initialization (init) time: *I-rate*);
3) once every performance-time control loop (perf time control rate, or *K-rate*);
4) once each sound sample of every control loop (perf time audio rate, or *A-rate*).

ORCHESTRA STATEMENT TYPES

An orchestra program in **Csound** is comprised of *orchestra header statements* which set various global parameters, followed by a number of *instrument blocks* representing different instrument types. An instrument block, in turn, is comprised of *ordinary statements* that set values, control the logical flow, or invoke the various signal processing subroutines that lead to audio output.

An *orchestra header statement* operates once only, at orchestra setup time. It is most commonly an assignment of some value to a *global reserved symbol*, e.g. sr = 20000. All orchestra header statements belong to a pseudo instrument 0, an *init* pass of which is run prior to all other instruments at score time 0. Any *ordinary statement* can serve as an orchestra header statement, eg. gifreq = cpspch(8.09), provided it is an init-time only operation.

An *ordinary statement* runs at either init time or performance time or both. Operations which produce a result formally run at the rate of that result (that is, at init time for I-rate results; at performance time for K- and A-rate results), with the sole exception of the **init** opcode (q.v.). Most **generators** and **modifiers**, however, produce signals that depend not only on the instantaneous value of their arguments but also on some preserved internal state. These performance-time units therefore have an implicit init-time component to set up that state. The run time of an operation which produces no result is apparent in the opcode.

Arguments are values that are sent to an operation. Most arguments will accept arithmetic expressions composed of constants, variables, reserved globals, value converters, arithmetic operations and conditional values; these are described below.

CONSTANTS AND VARIABLES

constants are floating point numbers, such as 1, 3.14159, or -73.45. They are available continuously and do not change in value.

variables are named cells containing numbers. They are available continuously and may be updated at one of the four update rates (setup only, I-rate, K-rate, or A-rate). I- and K-rate variables are scalars (i.e. they take on only one value at any given time) and are primarily used to store and recall controlling data, that is, data that changes at the note rate (for I-variables) or at the control rate (for K-variables). I- and K-variables are therefore useful for storing note parameter values, pitches, durations, slow-moving frequencies, vibratos, etc. A-variables, on the other hand, are arrays or vectors of information. Though renewed on the same perf-time control pass as K-variables, these array cells represent a finer resolution of time by dividing the control period into sample periods (see *ksmps* below). A-variables are used to store and recall data changing at the audio sampling rate (e.g. output signals of oscillators, filters, etc.).

A further distinction is that between local and global variables. **local** variables are private to a particular instrument, and cannot be read from or written into by any other instrument. Their values are preserved, and they may carry information from pass to pass (e.g. from initialization time to performance time) within a single instrument. Local variable names begin with the letter **p, i, k**, or **a**. The same local variable name may appear in two or more different instrument blocks without conflict.

global variables are cells that are accessible by all instruments. The names are either like local names preceded by the letter **g**, or are special reserved symbols. Global variables are used for broadcasting general values, for communicating between instruments (semaphores), or for sending sound from one instrument to another (e.g. mixing prior to reverberation).

Given these distinctions, there are eight forms of local and global variables:

type	when renewable	**Local**	**Global**
reserved symbols	permanent	--	rsymbol
score parameter fields	I-time	**p**number	--
init variables	I-time	**i**name	**gi**name
control signals	P-time, K-rate	**k**name	**gk**name
audio signals	P-time, A-rate	**a**name	**ga**name

where *rsymbol* is a special reserved symbol (e.g. **sr, kr**), *number* is a positive integer referring to a score statement pfield, and *name* is a string of letters and/or digits with local or global meaning. As might be inferred, score parameters are local I-variables whose values are copied from the invoking score statement just prior to the Init pass through an instrument.

VALUE CONVERTERS

ftlen(x)	(init-rate args only)
int(x)	(init- or control-rate args only)
frac(x)	(init- or control-rate args only)
dbamp(x)	(init- or control-rate args only)
i(x)	(control-rate args only)
abs(x)	(no rate restriction)
exp(x)	(no rate restriction)
log(x)	(no rate restriction)
sqrt(x)	(no rate restriction)
sin(x)	(no rate restriction)
cos(x)	(no rate restriction)
ampdb(x)	(no rate restriction)

where the argument within the parentheses may be an expression.

Value converters perform arithmetic translation from units of one kind to units of another. The result can then be a term in a further expression.

ftlen(x) returns the size (no. of points) of stored function table no. *x*.

int(x) returns the integer part of *x*.

frac(x) returns the fractional part of *x*.

dbamp(x) returns the decibel equivalent of the raw amplitude *x*.

i(x) returns an Init-type equivalent of the argument, thus permitting a K-time value to be accessed in at init-time or reinit-time, whenever valid.

abs(x) returns the absolute value of *x*.

exp(x) returns *e* raised to the *x*th power.

log(x) returns the natural log of *x* (*x* positive only).

sqrt(x) returns the square root of *x* (*x* non-negative).

sin(x) returns the sine of *x* (*x* in radians).

cos(x) returns the cosine of *x* (*x* in radians).

ampdb(x) returns the amplitude equivalent of the decibel value *x*.
Thus 60 db gives 1000, 66 db gives 2000, 72 db gives 4000,
78 db gives 8000, 84 db gives 16000 and 90 db gives 32000.

Note that for **log**, **sqrt**, and **ftlen** the argument value is restricted. Note also that **ftlen** will always return a power-of-2 value, i.e. the function table guard point (see F statement) is not included.

PITCH CONVERTERS

octpch(pch)	(init- or control-rate args only)
pchoct(oct)	(init- or control-rate args only)
cpspch(pch)	(init- or control-rate args only)
octcps(cps)	(init- or control-rate args only)
cpsoct(oct)	(no rate restriction)

where the argument within the parentheses may be a further expression.

These are really **value converters** with a special function of manipulating pitch data.

Data concerning pitch and frequency can exist in any of the following forms:

name	abbreviation
octave point pitch-class (8ve.pc)	**pch**
octave point decimal	**oct**
cycles per second	**cps**

The first two forms consist of a whole number, representing octave registration, followed by a specially interpreted fractional part. For **pch,** the fraction is read as two decimal digits representing the 12 equal-tempered pitch classes from.00 for C to.11 for B. For **oct,** the fraction is interpreted as a true decimal fractional part of an octave. The two fractional forms are thus related by the factor 100/12. In both forms, the fraction is preceded by a whole number octave index such that 8.00 represents Middle C,9.00 the C above, etc. Thus A440 can be represented alternatively by 440 (**cps**),8.09 (**pch**), 8.75 (oct), or 7.21 (**pch**), etc. Microtonal divisions of the pch semitone can be encoded by using more than two decimal places.

The mnemonics of the pitch conversion units are derived from morphemes of the forms involved, the second morpheme describing the source and the first morpheme the object (result). Thus

cpspch(8.09)

will convert the pitch argument 8.09 to its cps (or Hertz) equivalent, giving the value of 440. Since the argument is constant over the duration of the note, this conversion will take place at I-time, before any samples for the current note are produced. By contrast, the conversion

cpsoct(8.75 + K1)

which gives the value of A440 transposed by the octave interval K1 will repeat the calculation every, K-period since that is the rate at which K1 varies.

N.B. The conversion from **pch** or **oct** into **cps** is not a linear operation but involves an exponential process that could be time-consuming when executed repeatedly. **Csound** now uses a built-in table lookup to do this efficiently, even at audio rates.

ARITHMETIC OPERATIONS:

$-a$	
$+a$	
$a \;\&\&\; b$	(logical AND; not audio-rate)
$a \;\|\|\; b$	(logical OR; not audio-rate)
$a + b$	
$a - b$	
$a * b$	
$a \,/\, b$	

where the arguments *a* and *b* may be further expressions.

Arithmetic operators perform operations of change-sign (negate), don't-change-sign, logical AND logical OR, add, subtract, multiply and divide. Note that a value or an expression may fall between two of these operators, either of which could take it as its left or right argument, as in

$a + b * c.$

In such cases three rules apply:

1) * and / bind to their neighbors more strongly than + and -.
   Thus the above expression is taken as
   $a + (b * c),$
   with * taking *b* and *c* and then + taking *a* and $b * c$.
2) + and - bind more strongly than &&, which in turn is stronger than ||:
   $a \;\&\&\; b - c \;\|\|\; d$ is taken as $(a \;\&\&\; (b-c)) \;\|\|\; d$
3) When both operators bind equally strongly,
   the operations are done left to right:
   $a - b - c$ is taken as $(a - b) - c$.

Parentheses may be used as above to force particular groupings.

CONDITIONAL VALUES:

$(a > b \;\;?\;\; \mathrm{v1} : \mathrm{v2})$
$(a < b \;\;?\;\; \mathrm{v1} : \mathrm{v2})$
$(a \;>=\; b \;\;?\;\; \mathrm{v1} : \mathrm{v2})$
$(a \;<=\; b \;\;?\;\; \mathrm{v1} : \mathrm{v2})$
$(a \;==\; b \;\;?\;\; \mathrm{v1} : \mathrm{v2})$
$(a \;!=\; b \;\;?\;\; \mathrm{v1} : \mathrm{v2})$

where *a, b*, v1 and v2 may be expressions, but *a, b* not audio-rate.

In the above conditionals, *a* and *b* are first compared. If the indicated relation is true (*a* greater than *b*, *a* less than *b*, *a* greater than or equal to *b*, *a* less than or equal to *b*, *a* equal to *b*, *a* not equal to *b*), then the conditional expression has the value of v1; if the relation is false, the expression has the value of v2. (For convenience, a sole '=' will function as '= ='.) **NB**.: If v1 or v2 are expressions, these will be evaluated before the conditional is determined.

In terms of binding strength, all conditional operators (i. e., the relational operators ($>$,$<$, etc.), and ?, and :) are weaker than the arithmetic and logical operators (+, -, *, /, && and ||).

Example:

$(\mathrm{k1} < \mathrm{p5}/2 + \mathrm{p6} \;?\; \mathrm{k1} : \mathrm{p7})$

binds the terms p5/2 and p6. It will return the value k1 below this threshold, else the value p7.

EXPRESSIONS:

Expressions may be composed to any depth from the components shown above. Each part of an expression is evaluated at its own proper rate. For instance, if the terms within a sub-expression all change at the control rate or slower, the sub-expression will be evaluated only at the control rate; that result might then be used in an audio-rate evaluation. For example, in

```
k1 + abs(int(p5) + frac(p5) * 100/12 +  sqrt(k1))
```

the 100/12 would be evaluated at orch init, the p5 expressions evaluated at note I-time, and the remainder of the expression evaluated every k-period. The whole might occur in a unit generator argument position, or be part of an assignment statement .

DIRECTORIES and FILES:

Many generators and the **csound** command itself specify *filenames* to be read from or written to. These are optionally *full pathnames*, whose target directory is fully specified. When not fullpath, filenames are sought in several directories in order, depending on their type and on the setting of certain *environment variables*. The latter are optional, but they can serve to partition and organize the directories so that source files can be shared rather than duplicated in several user directories. The environment variables can define directories for soundfiles (SFDIR), sound samples (SSDIR), and sound analysis (SADIR). The search order is:

> Soundfiles being written are placed in SFDIR (if it exists), else the current directory.
> Soundfiles for reading are sought in the current directory, then SSDIR, then SFDIR.
> Analysis control files for reading are sought in the current directory, then SADIR.

NOMENCLATURE:

In **Csound** there are nine statement types, each of which provides a heading for the descriptive sections that follow in this chapter:

assignment statements	signal generator statements
orchestra header statements	signal modifier statements
instrument block statements	signal display statements
program control statements	soundfile access statements
duration control statements	

Throughout this document, opcodes are indicated in **boldface** and their argument and result mnemonics, when mentioned in the text, are given in *italics*. Argument names are generally mnemonic (*amp, phs*), and the result is denoted the letter *r*. Both are preceded by a type qualifier *i*, *k, a* or *x* (e.g. *kamp, iphs, ar*). The prefix *i* denotes scalar values valid at note Init time; prefixes *k* or *a* denote control (scalar) and audio (vector) values, modified and referenced continuously throughout performance (i.e. at every control period while the instrument is active). Arguments are *used* at the prefix-listed times; results are created at their listed times, then remain available for use as inputs elsewhere. The validity of inputs is defined by the following:

> arguments with prefix *i* must be valid at Init time;
> arguments with prefix *k* can be either control or Init values (which remain valid);
> arguments with prefix *a* must be vector inputs;
> arguments with prefix *x* may be either vector or scalar (the compiler will distinguish).

All arguments, unless otherwise stated, can be expressions whose results conform to the above. Most opcodes (such as **linen** and **oscil**) can be used in more than one mode, which one being determined by the prefix of the result symbol.

ASSIGNMENT STATEMENTS

```
ir      =       iarg
kr      =       karg
ar      =       xarg
kr      init    iarg
ar      init    iarg

ir      tival

ir      divz    ia, ib, isubst   (these not yet implemented)
kr      divz    ka, kb, ksubst
ar      divz    xa, xb, ksubst
```

= (simple assignment) - Put the value of the expression *iarg (karg, xarg)* into the named result. This provides a means of saving an evaluated result for later use.

init - Put the value of the I-time expression *iarg* into a K- or A-variable, i.e., initialize the result. Note that **init** provides the only case of an Init-time statement being permitted to write into a Perftime (K- or A-rate) result cell; the statement has no effect at Perf-time.

tival - Put the value of the instrument's internal "tie-in" flag into the named I-variable. Assigns 1 if this note has been 'tied' onto a previously held note (see **I** Statement); assigns 0 if no tie actually took place. (See also **tigoto**.)

divz - Whenever *b* is not zero, set the result to the value *a / b;* when *b* is zero, set it to the value of *subst* instead.

Example:

```
kcps    =       i2/3 + cpsoct(k2 + octpch(p5))
```

ORCHESTRA HEADER STATEMENTS

sr = n1
kr = n2
ksmps = n3
nchnls = n4

These statements are global value *assignments*, made at the beginning of an orchestra, before any instrument block is defined. Their function is to set certain *reserved symbol variables* that are required for performance. Once set, these reserved symbols can be used in expressions anywhere in the orchestra.

sr = (optional) - set sampling rate to *n1* samples per second per channel. The default value is 10000.

kr = (optional) - set control rate to *n2* samples per second. The default value is 1000.

ksmps = (optional) - set the number of samples in a Control Period to *n3*. **This value must equal sr/kr.** The default value is 10.

nchnls = (optional) - set number of channels of audio output to *n4*. (1 = mono, 2 = stereo, 4 = quadraphonic.) The default value is 1 (mono).

In addition, any **global variable** can be initialized by an *init-time assignment* anywhere before the first **instr** statement.

All of the above assignments are run as instrument 0 (i - pass only) at the start of real performance.

Example of header assignments:

```
        sr = 10000
        kr = 500
        ksmps = 20

gi1         =       sr/2.
ga          init    0
gitranspose =       octpch(.0l)
```

INSTRUMENT BLOCK STATEMENTS

```
instr     i, j, ...
.
.     < body
.       of
.       instrument >
.
endin
```

These statements delimit an instrument block. They must always occur in pairs.

instr - begin an instrument block defining instruments *i, j, ...*

i, j, ... must be numbers, not expressions. Any positive integer is legal, and in any order, but excessively high numbers are best avoided.

endin - end the current instrument block.

Note:

There may be any number of instrument blocks in an orchestra.

Instruments can be defined in any order (but they will always be both initialized and performed in ascending instrument number order).

Instrument blocks cannot be nested (i.e. one block cannot contain another).

PROGRAM CONTROL STATEMENTS

igoto	label
tigoto	label
kgoto	label
goto	label
if	ia R ib **igoto** label
if	ka R kb **kgoto** label
if	ia R ib **goto** label
timout	istrt, idur, label

where *label* is in the same instrument block and is not an expression, and where *R* is one of the Relational operators (>, <, >=, <=, ==, !=) (and = for convenience, see also under Conditional values).

These statements are used to control the order in which statements in an instrument block are to be executed. **I**-time and **P**-time passes can be controlled separately as follows:

igoto - During the I-time pass only, unconditionally transfer control to the statement labeled by *label.*

tigoto - similar to **igoto**, but effective only during an I-time pass at which a new note is being 'tied' onto a previously held note (see I Statement); no-op when a tie has not taken place. Allows an instrument to skip initialization of units according to whether a proposed tie was in fact successful (see also **tival**, **delay**).

kgoto - During the P-time passes only, unconditionally transfer control to the statement labeled by *label.*

goto - (combination of **igoto** and **kgoto**) Transfer control to *label* on every pass.

if...igoto - conditional branch at I-time, depending on the truth value of the logical expression "ia R ib". The branch is taken only if the result is true.

if...kgoto - conditional branch during P-time, depending on the truth value of the logical expression "ka R kb". The branch is taken only if the result is true.

if...goto - combination of the above. Condition tested on every pass.

timout - conditional branch during P-time, depending on elapsed note time. *istrt* and *idur* specify time in seconds. The branch to *label* will become effective at time *istrt,* and will remain so for just *idur* seconds. Note that **timout** can be reinitialized for multiple activation within a single note (see example next page).

Example:

```
if k3 > p5 + 10 kgoto next
```

reinit	label
rigoto	label
rireturn	

These statements permit an instrument to reinitialize itself during performance.

reinit - whenever this statement is encountered during a P-time pass, performance is temporarily suspended while a special Initialization pass, beginning at *label* and continuing to **rireturn** or **endin**, is executed. Performance will then be resumed from where it left off.

rigoto - similar to **igoto**, but effective only during a **reinit** pass (i.e., No-op at standard I-time). This statement is useful for bypassing units that are not to be reinitialized.

rireturn - terminates a **reinit** pass (i.e., No-op at standard I-time). This statement, or an **endin**, will cause normal performance to be resumed.

Example:

The following statements will generate an exponential control signal whose value moves from 440 to 880 exactly ten times over the duration p3.

```
reset:      timout      0, p3 /10, contin     ;after p3/10 seconds,
            reinit      reset                 ; reinit both timout
contin:     expon       440, p3/10,880        ; and expon
            riretum                           ; then resume perf
```

DURATION CONTROL STATEMENTS

ihold
turnoff

These statements permit the current note to modify its own duration.

ihold - this I-time statement causes a finite-duration note to become a ‘held’ note. It thus has the same effect as a negative p3 (see Score I-statement), except that p3 here remains positive and the instrument reclassifies itself to being held indefinitely. The note can be turned off explicitly with **turnoff**, or its space taken over by another note of the same instrument number (i.e. it is tied into that note). Effective at I-time only; no-op during a **reinit** pass.

turnoff - this P-time statement enables an instrument to turn itself off. Whether of finite duration or 'held', the note currently being performed by this instrument is immediately removed from the active note list. No other notes are affected.

Example:

The following statements will cause a note to terminate when a control signal passes a certain threshold (here the Nyquist frequency).

```
k1      expon  440, p3/10,880          ; begin gliss and continue
        if     k1 < sr/2 kgoto contin  ; until Nyquist detected
        turnoff                        ; then quit
contin:        a1     oscil   a1, k1, 1
```

MIDI CONVERTERS

ival	**notnum**	
ival	**veloc**	
icps	**cpsmidi**	
icps	**cpsmidib**	
kcps	**cpsmidib**	
ioct	**octmidi**	
ioct	**octmidib**	
koct	**octmidib**	
ipch	**pchmidi**	
ipch	**pchmidib**	
kpch	**pchmidib**	
iamp	**ampmidi**	iscal[, ifn]
kaft	**aftouch**	iscal
kchpr	**chpress**	iscal
kbend	**pchbend**	iscal
ival	**midictrl**	inum
kval	**midictrl**	inum

Get a value from the MIDI event that activated this instrument, or from a continuous MIDI controller, and convert it to a locally useful format.

INITIALIZATION

iscal - I-time scaling factor.

ifn (optional) - function table number of a normalized *translation* table, by which the incoming value is first interpreted. The default value is 0, denoting no translation.

inum - MIDI controller number.

PERFORMANCE

notnum, veloc - get the MIDI byte value (0 - 127) denoting the *note number* or *velocity* of the current event.

cpsmidi, octmidi, pchmidi - get the *note number* of the current MIDI event, expressed in **cps, oct,** or **pch** units for local processing.

cpsmidib, octmidib, pchmidib - get the *note number* of the current MIDI event, modify it by the current **pitch-bend** value, and express the result in **cps, oct,** or **pch** units. Available as an I-time value or as a continuous *ksig* value.

ampmidi - get the *velocity* of the current MIDI event, optionally pass it through a normalized translation table, and return an amplitude value in the range 0 - *iscal*.

aftouch, chpress, pchbend - get the current *after-touch*, *channel pressure*, or *pitch-bend* value for this channel, rescaled to the range 0 - *iscal*. Note that this access to *pitch-bend* data is independent of the MIDI pitch, enabling the value here to be used for any arbitrary purpose.

midictrl - get the current value (0 - 127) of a specified MIDI *controller*.

SIGNAL GENERATORS

kr	**line**	ia, idur1, ib
ar	**line**	ia, idur1, ib
kr	**expon**	ia, idur1, ib
ar	**expon**	ia, idur1, ib
kr	**linseg**	ia, idur1, ib[, idur2, ic[...]]
ar	**linseg**	ia, idur1, ib[, idur2, icI...]]
kr	**expseg**	ia, idur1, ib[, idur2, ic[...]]
ar	**expseg**	ia, idur1, ib[, idur2, ic[...]]

Output values *kr* or *ar* trace a straight line (exponential curve) or a series of line segments (exponential segments) between specified points.

INITIALIZATION

ia - starting value. Zero is illegal for exponentials.

ib, ic, etc. - value after *dur1* seconds, etc. For exponentials, must be non-zero and must agree in sign with *ia.*

idur1 - duration in seconds of first segment. A zero or negative value will cause all initialization to be skipped.

idur2, *idur3*, etc. - duration in seconds of subsequent segments. A zero or negative value will terminate the initialization process with the preceding point, permitting the last-defined line or curve to be continued indefinitely in performance. The default is zero.

PERFORMANCE

These units generate control or audio signals whose values can pass through 2 or more specified points. The sum of *dur* values may or may not equal the instrument's performance time: a shorter performance will truncate the specified pattern, while a longer one.will cause the last-defined segment to continue on in the same direction.

Example:

```
k2      expseg 440, p3/2,880, p3/2,440
```

This statement creates a control signal which moves exponentially from 440 to 880 and back, over the duration p3.

```
kr      phasor      kcps[, iphs]
ar      phasor      xcps[, iphs]
```

Produce a normalized moving phase value.

INITIALIZATION

iphs (optional) - initial phase, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to he skipped. The default value is zero.

PERFORMANCE

An internal phase is successively accumulated in accordance with the cps frequency to produce a moving phase value, normalized to lie in the range 0. <= phs < 1.

When used as the index to a **table** unit, this phase (multiplied by the desired function table length) will cause it to behave like an oscillator.

Note that **phasor** is a special kind of integrator, accumulating phase increments that represent frequency settings.

Example:

```
        phasor      1                       ; cycle once per second
kpch    table       k1 * 12, 1              ; through 12-note pch table
a1      oscil       p4, cpspch(kpch), 2     ; with continuous sound
```

ir	**table**	indx, ifn[, ixmode][, ixoff][, iwrap]
ir	**tablei**	indx, ifn[, ixmode][, ixoff][, iwrap]
kr	**table**	kndx, ifn[, ixmode][, ixoff][, iwrap]
kr	**tablei**	kndx, ifn[, ixmode][, ixoff][, iwrap]
ar	**table**	andx, ifn[, ixmode][, ixoff][, iwrap]
ar	**tablei**	andx, ifn[, ixmode][, ixoff][, iwrap]
kr	**oscil1**	idel, kamp, idur, ifn
kr	**oscil1i**	idel, kamp, idur, ifn

Table values are accessed by direct indexing or by incremental sampling.

INITIALIZATION

ifn - function table number. **tablei**, **oscil1i** require the extended guard point.

ixmode (optional) - ndx data mode. 0 = raw ndx, 1 = normalized (0 to 1). The default value is 0.

ixoff (optional) - amount by which ndx is to be offset. For a table with origin at center, use tablesize/2 (raw) or .5 (normalized). The default value is 0.

iwrap (optional) - wraparound ndx flag. 0 = nowrap (ndx < 0 treated as ndx=0; ndx > tablesize sticks at ndx=size), 1 = wraparound. The default value is 0.

idel - delay in seconds before **oscil1** incremental sampling begins.

idur - duration in seconds to sample through the **oscil1** table just once. A zero or negative value will cause all initialization to be skipped.

PERFORMANCE

table invokes table lookup on behalf of init, control or audio indices. These indices can be raw entry numbers (0,l,2...size - 1) or scaled values (0 to 1-e). Indices are first modified by the offset value then checked for range before table lookup (see *iwrap*). If ndx is likely to be full scale, or if interpolation is being used, the table should have an extended guard point. **table** indexed by a periodic phasor (see **phasor**) will simulate an oscillator.

oscil1 accesses values by sampling once through the function table at a rate determined by *idur*. For the first *idel* seconds, the point of scan will reside at the first location of the table; it will then begin moving through the table at a constant rate, reaching the end in another *idur* seconds; from that time on (i.e. after *idel* + *idur* seconds) it will remain pointing at the last location. Each value obtained from sampling is then multiplied by an amplitude factor *kamp* before being written into the result.

tablei and **oscil1i** are interpolating units in which the fractional part of ndx is used to interpolate between adjacent table entries. The smoothness gained by interpolation is at some small cost in execution time (see also **oscili**, etc.), but the interpolating and non-interpolating units are otherwise interchangeable. Note that when **tablei** uses a periodic index whose modulo *n* is less than the power of 2 table length, the interpolation process requires that there be an (*n* + 1)th table value that is a repeat of the 1st (see **F** statement in Score).

kr	**oscil**	kamp, kcps, ifn[, iphs]
kr	**oscili**	kamp, kcps, ifn[, iphs]
ar	**oscil**	xamp, xcps, ifn[, iphs]
ar	**oscili**	xamp, xcps, ifn[, iphs]
ar	**foscil**	xamp, kcps, kcar, kmod, kndx, ifn[, iphs]
ar	**foscili**	xamp, kcps, kcar, kmod, kndx, ifn[, iphs]

Table *ifn* is incrementally sampled modulo the table length and the value obtained is multiplied by *amp*.

INITIALIZATION

ifn - function table number. Requires a wrap-around guard point.

iphs (optional) - initial phase of sampling, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is 0.

PERFORMANCE

The **oscil** units output periodic control (or audio) signals consisting of the value of *kamp*(*xamp*)times the value returned from control rate (audio rate) sampling of a stored function table. The internal phase is simultaneously advanced in accordance with the *cps* input value. While the amplitude and frequency inputs to the K-rate **oscils** are scalar only, the corresponding inputs to the audio-rate **oscils** may each be either scalar or vector, thus permitting amplitude and frequency modulation at either sub-audio or audio frequencies.

foscil is a composite unit that effectively banks two **oscils** in the familiar Chowning FM setup, wherein the audio-rate output of one generator is used to modulate the frequency input of another (the "carrier"). Effective carrier frequency = *kcps* * *kcar*, and modulating frequency = *kcps* * *kmod*. For integral values of *kcar* and *kmod*, the perceived fundamental will be the minimum positive value of *kcps* * (*kcar* - *n* * *kmod*), *n = 1,1,2,...* The input *kndx* is the index of modulation (usually time-varying and ranging 0 to 4 or so) which determines the spread of acoustic energy over the partial positions given by n = 0,1,2,.., etc. *ifn* should point to a stored sine wave.

oscili and **foscili** differ from **oscil** and **foscil** respectively in that the standard procedure of using a truncated phase as a sampling index is here replaced by a process that interpolates between two successive lookups. Interpolating generators will produce a noticeably cleaner output signal, but they may take as much as twice as long to run. Adequate accuracy can also be gained without the time cost of interpolation by using large stored function tables of 2K, 4K or 8K points if the space is available.

Example:

```
k1      oscil   10, 5, 1             ; 5 cps vibrato
a1      oscil   5000, 440 + k1, 1    ; around A440 + -10 cps
```

ar1 [,ar2] **loscil** xamp, kcps, ifn[, ibas][, imod1,ibeg1,iend1][, imod2,ibeg2,iend2]

Read sampled sound (mono or stereo) from a table, with optional sustain and release looping.

INITIALIZATION

ifn - function table number, typically denoting an AIFF sampled sound segment with prescribed looping points. The source file may be mono or stereo.

ibas (optional) - base frequency in **cps** of the recorded sound. This optionally overrides the frequency given in the AIFF file, but is required if the file did not contain one. The default value is 0 (no override).

imod1, mod2 (optional) - play modes for the sustain and release loops. A value of 1 denotes normal looping, 2 denotes forward & backward looping, 0 denotes no looping. The default value (-1) will defer to the mode and the looping points given in the source file.

ibeg1, iend1, ibeg2, iend2 (optional, dependent on *mod1*, *mod2*) - begin and end points of the sustain and release loops. These are measured in **sample frames** from the beginning of the file, so will look the same whether the sound segment is monaural or stereo.

PERFORMANCE

loscil samples the ftable audio at a rate determined by *kcps*, then multiplies the result by *xamp*. The sampling increment for *kcps* is dependent on the table's base-note frequency *ibas,* and is automatically adjusted if the orchestra **sr** value differs from that at which the source was recorded. In this unit, ftable is always sampled with interpolation.

If sampling reaches the *sustain loop* endpoint and looping is in effect, the point of sampling will be modified and **loscil** will continue reading from within that loop segment. Once the instrument has received a *turnoff* signal (from the score or from a MIDI **noteoff** event), the next sustain endpoint encountered will be ignored and sampling will continue towards the *release loop* end-point, or towards the last sample (henceforth to zeros).

locscil is the basic unit for building a sampling synthesizer. Given a sufficient set of recorded piano tones, for example, this unit can resample them to simulate the missing tones. Locating the sound source nearest a desired pitch can be done via table lookup. Once a sampling instrument has begun, its *turnoff* point may be unpredictable and require an external *release* envelope; this is often done by gating the sampled audio with **linenr**, which will extend the duration of a turned-off instrument by a specific period while it implements a decay.

Example:

```
inum   notnum
icps   cpsmidi
iamp   ampmidi   3000, 1
ifno   table     inum, 2                    ;notnum to choose an audio sample
ibas   table     inum, 3
kamp   linenr    iamp, 0, .05, .01          ;at noteoff, extend by 50 millisecs
asig   loscil    kamp, icps, ifno, cpsoct(ibas/12. + 3)
```

ar **buzz** xamp, xcps, knh, ifn[, iphs]
ar **gbuzz** xamp, xcps, knh, klh, kr, ifn[, iphs]

Output is a set of harmonically related cosine partials.

INITIALIZATION

ifn - table number of a stored function containing (for **buzz**) a sine wave, or (for **gbuzz**) a cosine wave. In either case a large table of at least 8192 points is recommended.

iphs (optional) - initial phase of the fundamental frequency, expressed as a fraction of a cycle (0 to 1). A negative value will cause phase initialization to be skipped. The default value is zero.

PERFORMANCE

These units generate an additive set of harmonically related cosine partials of fundamental frequency *xcps*, and whose amplitudes are scaled so their summation peak equals *xamp*. The selection and strength of partials is determined by the following control parameters:

knh - total number of harmonics requested. Must be positive.

klh - lowest harmonic present. Can be positive, zero or negative. In **gbuzz** the set of partials can begin at any partial number and proceeds upwards; if *klh* is negative, all partials below zero will reflect as positive partials without phase change (since cosine is an even function), and will add constructively to any positive partials in the set.

kr - specifies the multiplier in the series of amplitude coefficients. This is a power series: if the *klh*th partial has a strength coefficient of A, the (*klh* + n)th partial will have a coefficient of A * (*kr* ** n), i.e. strength values trace an exponential curve. *kr* may be positive, zero or negative, and is not restricted to integers.

buzz and **gbuzz** are useful as complex sound sources in subtractive synthesis. **buzz** is a special case of the more general **gbuzz** in which *klh* = *kr* = 1; it thus produces a set of *knh* equal-strength harmonic partials, beginning with the fundamental. (This is a band-limited pulse train; if the partials extend to the Nyquist, i.e. *knh* = int (sr / 2 / fundamental freq.), the result is a real pulse train of amplitude *xamp*.) Although both *knh* and *klh* may be varied during performance, their internal values are necessarily integer and may cause "pops" due to discontinuities in the output; *kr*, however, can be varied during performance to good effect. Both **buzz** and **gbuzz** can be amplitude- and/or frequency-modulated by either control or audio signals.

N.B. These two units have their analogs in **GEN11**, in which the same set of cosines can be stored in a function table for sampling by an oscillator. Although computationally more efficient, the stored pulse train has a fixed spectral content, not a time-varying one as above.

ar	**adsyn**	kamod, kfmod, ksmod, ifilcod
ar	**pvoc**	ktimpnt, kfmod, ifilcod [, ispecwp]

Output is an additive set of individually controlled sinusoids, using either an oscillator bank or phase vocoder resynthesis.

INITIALIZATION

ifilcod - integer or character-string denoting a control-file derived from analysis of an audio signal. An integer denotes the suffix of a file *adsyn.m* or *pvoc.m*; a character-string (in double quotes) gives a filename, optionally a full pathname. If not fullpath, the file is sought first in the current directory, then in the one given by the environment variable SADIR (if defined). **adsyn** control contains breakpoint amplitude- and frequency-envelope values organized for oscillator resynthesis, while **pvoc** control contains similar data organized for fft resynthesis. Memory usage depends on the size of the files involved, which are read and held entirely in memory during computation but are shared by multiple calls (see also **lpread**).

ispecwp (optional) - if non-zero, attempts to preserve the spectral envelope while its frequency content is varied by *kfmod.* The default value is zero.

PERFORMANCE

adsyn synthesizes complex time-varying timbres through the method of additive synthesis. Any number of sinusoids, each individually controlled in frequency and amplitude, can be summed by high-speed arithmetic to produce a high-fidelity result.

Component sinusoids are described by a control file describing amplitude and frequency tracks in millisecond breakpoint fashion . Tracks are defined by sequences of 16-bit binary integers:

-1, time, amp, time, amp,...
-2, time, freq, time, freq,...

such as from hetrodyne filter analysis of an audio file. (For details see the Appendix on **hetro**.) The instantaneous amplitude and frequency values are used by an internal fixed-point oscillator that adds each active partial into an accumulated output signal. While there is a practical limit (currently 50) on the number of contributing partials, there is no restriction on their behavior over time. Any sound that can be described in terms of the behavior of sinusoids can be synthesized by **adsyn** alone.

Sound described by an **adsyn** control file can also be modified during re-synthesis. The signals *kamod*, *kfmod*, *ksmod* will modify the amplitude, frequency, and speed of contributing partials. These are multiplying factors, with *kfmod* modifying the cps frequency and *ksmod* modifying the *speed* with which the millisecond bread-point line-segments are traversed. Thus .7, 1.5, and 2 will give rise to a softer sound, a perfect fifth higher, but only half as long. The values 1,1,1 will leave the sound unmodified. Each of these inputs can be a control signal.

pvoc implements signal reconstruction using an fft-based phase vocoder. The control data stems from a precomputed analysis file with a known frame rate. The passage of time through this file is specified by *ktimpnt*, which represents the time in seconds. *ktimpnt* must always be positive, but can move forwards or backwards in time, be stationary or discontinuous, as a pointer into the analysis file. *kfmod* is a control-rate transposition factor: a value of 1 incurs no transposition, 1.5 transposes up a perfect fifth, and .5 down an octave.

This implementation of **pvoc** was written by Dan Ellis. It is based in part on the system of Mark Dolson, but the pre-analysis concept is new.

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

Audio output is a succession of sinusoid bursts initiated at frequency *xfund* with a spectral peak at *xform*. For *xfund* above 25 Hz these bursts produce a speech-like formant with spectral characteristics determined by the k-input parameters. For lower fundamentals this generator provides a special form of granular synthesis.

INITIALIZATION

iolaps - number of preallocated spaces needed to hold overlapping burst data. Overlaps are frequency dependent, and the space required depends on the maximum value of *xfund * kdur*. Can be over-estimated at no computation cost. Uses less than 50 bytes of memory per *iolap*.

ifna, *ifnb* - table numbers of two stored functions. The first is a sine table for sineburst synthesis (size of at least 4096 recommended). The second is a rise shape, used forwards and backwards to shape the sineburst rise and decay; this may be linear (GEN07) or perhaps a sigmoid (GEN19).

itotdur - total time during which this **fof** will be active. Normally set to p3. No new sineburst is created if it cannot complete its *kdur* within the remaining *itotdur*.

iphs (optional) - initial phase of the fundamental, expressed as a fraction of a cycle (0 to 1). The default value is 0.

ifmode (optional) - formant frequency mode. If zero, each sineburst keeps the *xform* frequency it was launched with. If non-zero, each is influenced by *xform* continuously. The default value is 0.

PERFORMANCE

xamp - peak amplitude of each sineburst, observed at the true end of its rise pattern. The rise may exceed this value given a large bandwidth (say, Q < 10) and/or when the bursts are overlapping.

xfund - the fundamental frequency (in Hertz) of the impulses that create new sinebursts.

xform - the formant frequency, i.e. freq of the sinusoid burst induced by each *xfund* impulse. This frequency can be fixed for each burst or can vary continuously (see *ifmode*).

koct - octaviation index, normally zero. If greater than zero, lowers the effective *xfund* frequency by attenuating odd-numbered sinebursts. Whole numbers are full octaves, fractions transitional.

kband - the formant bandwidth (at -6dB), expressed in Hz. The bandwidth determines the rate of exponential decay throughout the sineburst, before the enveloping described below is applied.

kris, kdur, kdec - rise, overall duration, and decay times (in seconds) of the sinusoid burst. These values apply an enveloped duration to each burst, in similar fashion to a Csound **linen** generator but with rise and decay shapes derived from the *ifnb* input. *kris* inversely determines the skirtwidth (at -40 dB) of the induced formant region. *kdur* affects the density of sineburst overlaps, and thus the speed of computation. Typical values for vocal imitation are .003,.02,.007.

Csound's **fof** generator is loosely based on Michael Clarke's C-coding of IRCAM's CHANT program (Xavier Rodet et al.). Each **fof** produces a single formant, and the output of four or more of these can be summed to produce a rich vocal imitation. **fof** synthesis is a special form of granular synthesis, and this implementation aids transformation between vocal imitation and granular textures. Computation speed depends on *kdur, xfund,* and the density of any overlaps.

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

Audio output is a naturally decaying plucked string or drum sound based on the Karplus-Strong algorithms.

INITIALIZATION

icps - intended pitch value in cps, used to set up a buffer of 1 cycle of audio samples which will be smoothed over time by a chosen decay method. *icps* normally anticipates the value of *kcps*, but may be set artificially high or low to influence the size of the sample buffer.

ifn - table number of a stored function used to initialize the cyclic decay buffer. If *ifn* = 0, a random sequence will be used instead.

imeth - method of natural decay. There are six, some of which use parameters values that follow.

1 - simple averaging. A simple smoothing process, uninfluenced by parameter values.
2 - stretched averaging. As above, with smoothing time stretched by a factor of *iparm1* (>= 1).
3 - simple drum. The range from pitch to noise is controlled by a 'roughness factor' in *iparm1* (0 to 1). Zero gives the plucked string effect, while 1 reverses the polarity of every sample (octave down, odd harmonics). The setting .5 gives an optimum snare drum.
4 - stretched drum. Combines both roughness and stretch factors. *iparm1* is roughness (0 to 1), and *iparm2* the stretch factor (>= 1).
5 - weighted averaging. As method 1, with *iparm1* weighting the current sample (the status quo) and *iparm2* weighting the previous adjacent one. *iparm1* + *iparm2* must be <= 1.
6 - 1st order recursive filter, with coefs .5. Unaffected by parameter values.

iparm1, *iparm2* (optional) - parameter values for use by the smoothing algorithms (above). The default values are both 0.

PERFORMANCE

An internal audio buffer, filled at I-time according to *ifn*, is continually resampled with periodicity *kcps* and the resulting output is multiplied by *kamp*. Parallel with the sampling, the buffer is smoothed to simulate the effect of natural decay.

Plucked strings (1,2,5,6) are best realized by starting with a random noise source, which is rich in initial harmonics. Drum sounds (methods 3,4) work best with a flat source (wide pulse), which produces a deep noise attack and sharp decay.

The original Karplus-Strong algorithm used a fixed number of samples per cycle, which caused serious quantization of the pitches available and their intonation. This implementation resamples a buffer at the exact pitch given by *kcps*, which can be varied for vibrato and glissando effects. For low values of the orch sampling rate (e.g. *sr* = 10000), high frequencies will store only very few samples (*sr* / *icps*). Since this may cause noticeable noise in the resampling process, the internal buffer has a minimum size of 64 samples. This can be further enlarged by setting *icps* to some artificially lower pitch.

kr	**rand**	xamp[, iseed]
kr	**randh**	kamp, kcps[, iseed]
kr	**randi**	kamp, kcps[, iseed]
ar	**rand**	xamp[, iseed]
ar	**randh**	xamp, xcps[, iseed]
ar	**randi**	xamp, xcps[, iseed]

Output is a controlled random number series between +*amp* and -*amp*

INITIALIZATION

iseed (optional) - seed value for the recursive psuedo-random formula. A value between 0 and +1 will produce an initial output of *kamp* * *iseed* A negative value will cause seed re-initialization to be skipped. The default seed value is .5.

PERFORMANCE

The internal psuedo-random formula produces values which are uniformly distributed over the range *kamp* to -*kamp*. **rand** will thus generate uniform white noise with an R.M.S value of *kamp / root 2*.

The remaining units produce band-limited noise: the cps parameters permit the user to specify that new random numbers are to be generated at a rate less than the sampling or control frequencies. **randh** will hold each new number for the period of the specified cycle; **randi** will produce straightline interpolation between each new number and the next.

Example:

```
i1    =       octpch(p5)     ; center pitch, to be modified
k1    randh   1,10           ;10 time/sec by random displacements up to 1 octave
a1    oscil   5000, cpsoct(i1+ k1), 1
```

SIGNAL MODIFIERS

kr	**linen**	kamp, irise, idur, idec
ar	**linen**	xamp, irise, idur, idec
kr	**linenr**	kamp, irise, idec, iatdec
ar	**linenr**	xamp, irise, idec, iatdec
kr	**envlpx**	kamp, irise, idur, idec, ifn, iatss, iatdec[, ixmod]
ar	**envlpx**	xamp, irise, idur, idec, ifn, iatss, iatdec[, ixmod]

linen - apply a straight line rise and decay pattern to an input amp signal.
linenr - apply a straight line rise, then an exponential decay while the note is extended in time.
envlpx - apply an envelope consisting of 3 segments: 1) stored function rise shape, 2) modified exponential "pseudo steady state", 3) exponential decay

INITIALIZATION

irise - rise time in seconds. A zero or negative value signifies no rise modification.

idur - overall duration in seconds. A zero or negative value will cause initialization to be skipped.

idec - decay time in seconds. Zero means no decay. An idec > *idur* will cause a truncated decay.

ifn - function table number of stored rise shape with extended guard point.

iatss - attenuation factor, by which the last value of the **envlpx** rise is modified during the note's pseudo "steady state." A factor > l causes an exponential growth, and < l an exponential decay. A 1 will maintain a true steady state at the last rise value. Note that this attenuation is not by fixed rate (as in a piano), but is sensitive to a note's duration. However, if *iatss* is negative (or if "steady state" < 4 k-periods) a fixed attenuation rate of *abs*(*iatss*) per second will be used. 0 is illegal.

iatdec - attenuation factor by which the closing "steady state" value is reduced exponentially over the decay period. This value must be positive and is normally of the order of .01. A large or excessively small value is apt to produce a cutoff which is audible. A zero or neg value is illegal.

ixmod (optional, between +- .9 or so) - exponential curve modifier, influencing the "steepness" of the exponential trajectory during the "steady state." Values less than zero will cause an accelerated growth or decay towards the target (e.g. *subito piano*). Values greater than zero will cause a retarded growth or decay. The default value is zero (unmodified exponential).

PERFORMANCE

Rise modifications are applied for the first *irise* seconds, and decay from time *idur* - *idec*. If these periods are separated in time there will be a "steady state" during which *amp* will be unmodified (**linen**) or modified by the first exponential pattern (**envlpx**). If **linen** rise and decay periods overlap then both modifications will be in effect for that time; in **envlpx** that will cause a truncated decay. If the overall duration *idur* is exceeded in performance, the final decay will continue on in the same direction, going negative for **linen** but tending asymptotically to zero in **envlpx**.

linenr is unique within Csound in containing a **note-off sensor** and **release time extender.** When it senses either a score event termination or a MIDI noteoff, it will immediately extend the performance time of the current instrument by *idec* seconds, then execute an exponential decay towards the factor *iatdec*. For two or more units in an instrument, extension is by the greatest *idec*.

kr	**port**	ksig, ihtim[, isig]
ar	**tone**	asig, khp[, istor]
ar	**atone**	asig, khp[, istor]
ar	**reson**	asig, kcf, kbw[, iscl, istor]
ar	**areson**	asig, kcf, kbw[, iscl, istor]

A control or audio signal is modified by a low- or band-pass recursive filter with variable frequency response.

INITIALIZATION

isig - initial (i.e. previous) value for internal feedback. The default value is 0.

istor - initial disposition of internal data space. Since filtering incorporates a feedback loop of previous output, the initial status of the storage space used is significant. A zero value will clear the space; a non-zero value will allow previous information to remain. The default value is 0.

iscl - coded scaling factor for resonators. A value of 1 signifies a peak response factor of 1, i.e. all frequencies other than *kcf* are attenuated in accordance with the (normalized) response curve. A value of 2 raises the response factor so that its overall RMS value equals 1. (This intended equalization of input and output power assumes all frequencies are physically present; hence it is most applicable to white noise.) A zero value signifies no scaling of the signal, leaving that to some later adjustment (e.g. see **balance**). The default value is 0.

PERFORMANCE

port applies portamento to a step-valued control signal. At each new step value, *ksig* is low-pass filtered to move towards that value at a rate determined by *ihtim*. *ihtim* is the "half-time" of the function (in seconds), during which the curve will traverse half the distance towards the new value, then half as much again, etc., theoretically never reaching its asymptote.

tone implements a first-order recursive low-pass filter in which the variable *khp* (in cps) determines the response curve's half -power point. Half power is defined as peak power / root 2.

reson is a second-order filter in which *kcf* controls the center frequency, or cps position of the peak response, and *kbw* controls its bandwidth (the cps difference between the upper and lower half -power points).

atone, **areson** are filters whose transfer functions are the complements of **tone** and **reson**. **atone** is thus a form of high-pass filter and **areson** a notch filter whose transfer functions represent the "filtered out" aspects of their complements. Note, however, that power scaling is not normalized in **atone**, **areson**, but remains the true complement of the corresponding unit. Thus an audio signal, filtered by parallel matching **reson** and **areson** units, would under addition simply reconstruct the original spectrum. This property is particularly useful for controlled mixing of different sources (e.g., see **lpreson**).

Complex response curves such as those with multiple peaks can be obtained by using a bank of suitable filters in series. (The resultant response is the product of the component responses.) In such cases, the combined attenuation may result in a serious loss of signal power, but this can be regained by the use of **balance**.

krmsr,krmso,kerr,kcps	**lpread**	ktimpnt, ifilcod[, inpoles][, ifrmrate]
ar	**lpreson**	asig
ar	**lpfreson**	asig, kfrqratio

These units, used as a read/reson pair, use a control file of time-varying filter coefficients to dynamically modify the spectrum of an audio signal.

INITIALIZATION

ifilcod - integer or character-string denoting a control-file (reflection coefficients and four parameter values) derived from n-pole linear predictive spectral analysis of a source audio signal. An integer denotes the suffix of a file *lp.m;* a character-string (in double quotes) gives a filename, optionally a full pathname. If not fullpath, the file is sought first in the current directory, then in that of the environment variable SADIR (if defined). Memory usage depends on the size of the file, which is held entirely in memory during computation but shared by multiple calls (see also **adsyn, pvoc**).

inpoles, *ifrmrate* (optional) - number of poles, and frame rate per second in the lpc analysis. These arguments are required only when the control file does not have a header; they are ignored when a header is detected. The default value for both is zero.

PERFORMANCE

lpread accesses a control file of time-ordered information frames, each containing n-pole filter coefficients derived from linear predictive analysis of a source signal at fixed time intervals (e.g. 1/100 of a second), plus four parameter values:

krmsr - root-mean-square (rms) of the residual of analysis,
krmso - rms of the original signal,
kerr - the normalized error signal,
kcps - pitch in cps.

lpread gets its values from the control file according to the input value *ktimpnt* (in seconds). If *ktimpnt* proceeds at the analysis rate, time-normal synthesis will result; proceeding at a faster, slower, or variable rate will result in time-warped synthesis. At each K-period, **lpread** interpolates between adjacent frames to more accurately determine the parameter values (presented as output) and the filter coefficient settings (passed internally to a subsequent **lpreson**).

The error signal *kerr* (between 0 and 1) derived during predictive analysis reflects the deterministic/random nature of the analyzed source. This will emerge low for pitched (periodic) material and higher for noisy material. The transition from voiced to unvoiced speech, for example, produces an error signal value of about .3. During synthesis, the error signal value can be used to determine the nature of the **lpreson** driving function: for example, by arbitrating between pitched and non-pitched input, or even by determining a mix of the two. In normal speech resynthesis, the pitched input to **lpreson** is a wideband periodic signal or pulse train derived from a unit such as **buzz**, and the nonpitched source is usually derived from **rand**. However, any audio signal can be used as the driving function, the only assumption of the analysis being that it has a flat response.

lpfreson is a formant shifted **lpreson**, in which *kfrqratio* is the (cps) ratio of shifted to original formant positions. This permits synthesis in which the source object changes its apparent acoustic size. **lpfreson** with *kfrqratio* = 1 is equivalent to **lpreson**.

Generally, **lpreson** provides a means whereby the time-varying content and spectral shaping of a composite audio signal can be controlled by the dynamic spectral content of another. There can be any number of **lpread/lpreson** (or **lpfreson**) pairs in an instrument or in an orchestra; they can read from the same or different control files independently.

```
kr      rms         asig[, ihp, istor]
nr      gain        asig, krms[, ihp, istor]
ar      balance     asig, acomp[, ihp, istor]
```

The rms power of *asig* can be interrogated, set, or adjusted to match that of a comparator signal.

INITIALIZATION

ihp (optional) - half-power point (in cps) of a special internal low-pass filter. The default value is 10.

istor (optional) - initial disposition of internal data space (see **reson**). The default value is 0.

PERFORMANCE

rms output values kr will trace the **rms** value of the audio input *asig*. This unit is not a signal modifier, but functions rather as a signal power-guage.

gain provides an amplitude modification of *asig* so that the output *ar* has **rms** power equal to *krms*. **rms** and **gain** used together (and given matching *ihp* values) will provide the same effect as **balance**.

balance outputs a version of *asig*, amplitude-modified so that its rms power is equal to that of a comparator signal *acomp*. Thus a signal that has suffered loss of power (eg., in passing through a filter bank) can be restored by matching it with, for instance, its own source. It should be noted that **gain** and **balance** provide amplitude modification only - output signals are not altered in any other respect.

Example:

```
asrc    buzz        10000,440, sr/440, 1        ; band-limited pulse train
a1      reson       asrc, 1000,100              ; sent through
a2      reson       a1,3000,500                 ; 2 filters
afin    balance     a2, asrc                    ; then balanced with source
```

kr	**downsamp**	asig[, iwlen]
ar	**upsamp**	ksig
ar	**interp**	ksig[, istor]
kr	**integ**	ksig[, istor]
ar	**integ**	asig[, istor]
kr	**diff**	ksig[, istor]
ar	**diff**	asig[, istor]
kr	**samphold**	xsig, kgate[, ival, ivstor]
ar	**samphold**	asig, xgate[, ival, ivstor]

Modify a signal by up- or down-sampling, integration, and differentiation.

INITIALIZATION

iwlen (optional) - window length in samples over which the audio signal is averaged to determine a downsampled value. Maximum length is *ksmps*; 0 and 1 imply no window averaging. The default value is 0.

istor (optional) - initial disposition of internal save space (see **reson**). The default value is 0.

ival, *ivstor* (optional) - controls initial disposition of internal save space. If *ivstor* is zero the internal “hold" value is set to *ival* ; else it retains its previous value. Defaults are 0,0 (i.e. init to zero).

PERFORMANCE

downsamp converts an audio signal to a control signal by downsampling. It produces one kval for each audio control period. The optional window invokes a simple averaging process to suppress foldover.

upsamp, **interp** convert a *control* signal to an *audio* signal. The first does it by simple repetition of the kval, the second by linear interpolation between successive kvals. **upsamp** is a slightly more efficient form of the assignment, ‘asig = ksig’.

integ, **diff** perform *integration* and *differentiation* on an input control signal or audio signal. Each is the converse of the other, and applying both will reconstruct the original signal. Since these units are special cases of low-pass and high-pass filters, they produce a scaled (and phase shifted) output that is frequency-dependent. Thus **diff** of a sine produces a cosine, with amplitude *2 * sin(pi * cps /* **sr***)* that of the original (for each component partial); **integ** will inversely affect the magnitudes of its component inputs. With this understanding, these units can provide useful signal modification.

samphold performs a sample-and-hold operation on its input according to the value of *gate*. If *gate > 0*, the input samples are passed to the output; If *gate* <= 0, the last output value is repeated. The controlling *gate* can be a constant, a control signal, or an audio signal.

Example:

```
asrc   buzz       10000,440,20, 1   ; band-limited pulse train
adif   diff       asrc              ; emphasize the highs
anew   balance    adif, asrc        ;       but retain the power
agate  reson      asrc,0,440        ; use a lowpass of the original
asamp  samphold   anew, agate       ;       to gate the new audiosig
aout   tone       asamp,100         ; smooth out the rough edges
```

```
ar      delayr      idlt[, istor]
        delayw      asig
ar      delay       asig, idlt[, istor]
ar      delay1      asig[, istor]
```

A signal can be read from or written into a delay path, or it can be automatically delayed by some time interval.

INITIALIZATION

idlt - requested delay time in seconds. This can be as large as available memory will permit. The space required for *n* seconds of delay is *4n* * **sr** bytes. It is allocated at the time the instrument is first initialized, and returned to the pool at the end of a score section.

istor (optional) - initial disposition of delay-loop data space (see **reson**). The default value is 0.

PERFORMANCE

delayr reads from an automatically established digital delay line, in which the signal retrieved has been resident for *idlt* seconds. This unit must be paired with and precede an accompanying **delayw** unit. Any other **Csound** statements can intervene.

delayw writes *asig* into the delay area established by the preceding **delayr** unit. Viewed as a pair, these two units permit the formation of modified feedback loops, etc. However, there is a lower bound on the value of *idlt*, which must be at least 1 control period (or 1/**kr**).

delay is a composite of the above two units, both reading from and writing into its own storage area. It can thus accomplish signal time-shift, although modified feedback is not possible. There is no minimum delay period.

delay1 is a special form of delay that serves to delay the audio signal *asig* by just one sample. It is thus functionally equivalent to "**delay** asig, 1/**sr**" but is more efficient in both time and space. This unit is particularly useful in the fabrication of generalized non-recursive filters.

Example:

```
        tigoto  contin          ; except on a tie,
a2      delay   a1, .05, 0      ; begin 50 msec clean delay of sig
contin:
```

```
ar      deltap      kdlt
ar      deltapi     xdlt
```

Tap a delay line at variable offset times.

PERFORMANCE

These units can tap into a **delayr/delayw** pair, extracting delayed audio from the *idlt* seconds of stored sound. There can be any number of **deltap** and/or **deltapi** units between a read/write pair. Each receives an audio tap with no change of original amplitude.

deltap extracts sound by reading the stored samples directly; **deltapi** extracts sound by interpolated readout. By interpolating between adjacent stored samples **deltapi** represents a particular delay time with more accuracy, but it will take about twice as long to run.

The arguments *kdlt*, *xdlt* specify the tapped delay time in seconds. Each can range from 1 Control Period to the full delay time of the read/write pair; however, since there is no internal check for adherence to this range, the user is wholly responsible. Each argument can be a constant, a variable, or a time-varying signal; the *xdlt* argument in **deltapi** implies that an audio-varying delay is permitted there.

These units can provide multiple delay taps for arbitrary delay path and feedback networks. They can deliver either constant-time or time-varying taps, and are useful for building chorus effects, harmonizers, and doppler shifts. Constant-time delay taps (and some slowly changing ones) do not need interpolated readout; they are well served by **deltap**. Medium-paced or fast varying *dlt's*, however, will need the extra services of **deltapi**.

N.B. K-rate delay times are not internally interpolated, but rather lay down stepped time-shifts of audio samples; this will be found quite adequate for slowly changing tap times. For medium to fastpaced changes, however, one should provide a higher resolution audio-rate timeshift as input.

Example:

```
asource     buzz    1, 440, 20, 1
atime       linseg  1, p3/2,.01, p3/2,1     ; trace a distance in secs
ampfac      =       1/atime/atime           ;     and calc an amp factor
adump       delayr  1                       ; set maximum distance
amove       deltapi atime                   ; move sound source past
            delays  asource                 ;     the listener
            out     amove * ampfac
```

```
ar      comb        asig, krvt, ilpt[, istor]
ar      alpass      asig, krvt, ilpt[, istor]
ar      reverb      asig, krvt[, istor]
```

An input signal is reverberated for *krvt* seconds with "colored" (**comb**), flat (**alpass**), or "natural room" (**reverb**) frequency response.

INITIALIZATION

ilpt - loop time in seconds, which determines the "echo density" of the reverberation. This in turn characterizes the "color" of the **comb** filter whose frequency response curve will contain *ilpt* * **sr**/2 peaks spaced evenly between 0 and **sr**/2 (the Nyquist frequency). Loop time can be as large as available memory will permit. The space required for an *n* second loop is *4n* * **sr** bytes. **comb** and **alpass** delay space is allocated and returned as in **delay**.

istor (optional) - initial disposition of delay-loop data space (cf. **reson**). The default value is 0.

PERFORMANCE

These filters reiterate input with an echo density determined by loop time *ilpt*. The attenuation rate is independent and is determined by *krvt*, the reverberation time (defined as the time in seconds for a signal to decay to 1/1000, or 60dB down from its original amplitude). Output from a **comb** filter will appear only after *ilpt* seconds; **alpass** output will begin to appear immediately.

A standard **reverb** unit is composed of four **comb** filters in parallel followed by two **alpass** units in series. Looptimes are set for optimal "natural room response." Core storage requirements for this unit are proportional only to the sampling rate, each unit requiring approximately 3K words for every 10KC. The **comb**, **alpass**, **delay**, **tone** and other **Csound** units provide the means for experimenting with alternate reverberator designs

Since output from the standard **reverb** will begin to appear only after 1/20 second or so of delay, and often with less than three-fourths of the original power, it is normal to output both the source and the reverberated signal. Also, since the reverberated sound will persist long after the cessation of source events, it is normal to put **reverb** in a separate instrument to which sound is passed via a *global variable*, and to leave that instrument running throughout the performance.

Example:

```
ga1     init    0                       ; init an audio receiver/mixer

        instr   1                       ; instr (there may be many copies)
a1      oscili  8000, cpspch(p5), 1     ; generate a source signal
        out     a1                      ; output the direct sound
ga1     =       ga1 + a1                ; and add to audio receiver
        endin

        instr   99                      ; (highest instr number executed last)
a3      reverb  ga1, 1.5                ; reverberate whatever is in ga1
        out     a1                      ; and output the result
ga1     =       0                       ; empty the receiver for the next pass
        endin
```

OPERATIONS USING SPECTRAL DATA-TYPES

dsig	**octdown**	xsig, iocts, isamps[, idisprd]
wsig	**noctdft**	dsig, iprd, ifrqs, iq[, ihann, idbout, idsines]
wsig	**specscal**	wsigin, ifscale, ifthresh
wsig	**specaddm**	wsig1, wsig2[, imul2]
wsig	**specdiff**	wsigin
wsig	**specaccm**	wsigin
wsig	**specfilt**	wsigin, ifhtim
	specdisp	wsig, iprd[, iwtflg]
ksum	**specsum**	wsig[, interp]

These units generate and process non-standard signal data types, such as down-sampled time-domain control signals and audio signals, and their frequency-domain (spectral) representations. The new data types (d-, w-) are self-defining, and the contents are not processable by any other Csound units. These unit generators are experimental, and subject to change between releases; they will also be joined by others later. Their inclusion here is to offer the user some initial experience in spectral data processing.

INITIALIZATION

idisprd (optional) - if non-zero, display the output every *idisprd* seconds. The default value is 0 (no display).

ihann, idbout, idsines (optional) - if non-zero, then respectively: apply a hanning window to the input; convert the output magnitudes to dB; display the windowed sinusoids used in DFT filtering. The default values are 0,0,0 (rectangular window, magnitude output, no sinusoid display).

imul2 (optional) - if non-zero, scale the *wsig2* magnitudes before adding. The default value is 0.

iwtflg (optional) - wait flag. If non-zero, hold each display until released by the user. The default value is 0 (no wait).

interp (optional) - if non-zero, interpolate the output signal *ksum*. The default value is 0 (repeat the signal value between changes).

PERFORMANCE

octdown - put signal *asig* or *ksig* through *iocts* successive applications of octave decimation and downsampling, and preserve *isamps* down-sampled values in each octave. Optionally display the composite buffer every *idisprd* seconds.

noctdft - generate a constant-Q, exponentially-spaced DFT across all octaves of the multiply-downsampled input *dsig.* Every *iprd* seconds, each octave of *dsig* is optionally windowed (*ihann* non-zero), filtered (using *ifrqs* parallel filters per octave, exponentially spaced, and with frequency/bandwidth Q of *iq*), and the output magnitudes optionally converted to dB (*idbout* non-zero). This unit produces a self-defining spectral datablock *wsig*, whose characteristics are readable by any units that receive it as input, and for which it becomes the template for output.

specscal - scale an input spectral datablock with spectral envelopes. Function tables *ifthresh* and *ifscale* are initially sampled across the (logarithmic) frequency space of the input spectrum; then each time a new input spectrum is sensed the sampled values are used to scale each of its magnitude channels as follows: if *ifthresh* is non-zero, each magnitude is reduced by its corresponding table-value (to not less than zero); then each magnitude is rescaled by the corresponding *ifscale* value, and the resulting spectrum written to *wsig*.

specaddm - do a weighted add of two input spectra. For each channel of the two input spectra, the two magnitudes are combined and written to the output according to: magout = mag1in + mag2in * *imul2*. The operation is performed whenever the input *wsig1* is sensed to be new. This unit will (at Initialization) verify the consistency of the two spectra (equal size, equal period, equal mag types).

specdiff - find the positive difference values between consecutive spectral frames. At each new frame of *wsigin*, each magnitude value is compared with its predecessor, and the positive changes written to the output spectrum. This unit is useful as an energy onset detector.

specaccm - accumulate the values of successive spectral frames. At each new frame of *wsigin*, the accumulations-to-date in each magnitude track are written to the output spectrum. This unit thus provides a running *histogram* of spectral distribution.

specfilt - filter each channel of an input spectrum. At each new frame of *wsigin*, each magnitude value is injected into a 1st-order lowpass recursive filter, whose half-time constant has been initially set by sampling the ftable *ifhtim* across the (logarithmic) frequency space of the input spectrum. This unit effectively applies a *persistence* factor to the data occurring in each spectral channel, and is useful for simulating the *energy integration* that occurs during auditory perception.

specdisp - display the magnitude values of spectrum *wsig* every *iprd* seconds (rounded to some integral number of *wsig's* originating *iprd*).

specsum - sum the magnitudes across all channels of the spectrum. At each new frame of *wsig*, the magnitudes are summed and released as a scalar *ksum* signal. Between frames, the output is either repeated or interpolated at the K-rate. This unit produces a k-signal summation of the magnitudes present in the spectral data, and is thereby a running measure of its moment-to-moment overall strength.

Example:

```
asig    in                                          ; get external audio
dsamp   octdown    asig, 6, 180, 0                  ; downsample in 6 octaves
wsig1   noctdft    dsamp, .02, 12, 33, 0, 1, 1      ; and calc 72-point dft (dB)
wsig2   specdiff   wsig1                            ; sense onsets
wsig3   specfilt   wsig2, 2                         ; & absorb slowly
        specdisp   wsig1, .1                        ; display all spectra
        specdisp   wsig2, .1
        specdisp   wsig3, .1
kstren  specsum    wsig3,  1                        ; sum final mags, and ksmooth
```

SENSING & CONTROL

ktemp **tempest** kin, iprd, imindur, imemdur, ihp, ithresh, ihtim, ixfdbak, istartempo, ifn[, idisprd, itweek]

Estimate the tempo of beat patterns in a control signal.

INITIALIZATION

iprd - period between analyses (in seconds). Typically about .02 seconds.

imindur - minimum duration (in seconds) to serve as a unit of tempo. Typically about .2 seconds.

imemdur - duration (in seconds) of the *kin* short-term memory buffer which will be scanned for periodic patterns. Typically about 3 seconds.

ihp - half-power point (in cps) of a low-pass filter used to smooth input *kin* prior to other processing. This will tend to suppress activity that moves much faster. Typically 2 cps.

ithresh - loudness threshold by which the low-passed *kin* is center-clipped before being placed in the short-term buffer as tempo-relevant data. Typically at the noise floor of the incoming data.

ihtim - half-time (in seconds) of an internal forward-masking filter that masks new *kin* data in the presence of recent, louder data. Typically about .005 seconds.

ixfdbak - proportion of this unit's *anticipated value* to be mixed with the incoming *kin* prior to all processing. Typically about .3.

istartempo - initial tempo (in beats per minute). Typically 60.

ifn - table number of a stored function (drawn left-to-right) by which the short-term memory data is attenuated over time.

idisprd (optional) - if non-zero, display the short-term past and future buffers every *idisprd* seconds (normally a multiple of *iprd*). The default value is 0 (no display).

itweek (optional) - fine-tune adjust this unit so that it is stable when analyzing events controlled by its own output. The default value is 1 (no change).

PERFORMANCE

tempest examines *kin* for amplitude periodicity, and estimates a current tempo. The input is first low-pass filtered, then center-clipped, and the residue placed in a short-term memory buffer (attenuated over time) where it is analyzed for periodicity using a form of autocorrelation. The period, expressed as a *tempo* in beats per minute, is output as *ktemp*. The period is also used internally to make predictions about future amplitude patterns, and these are placed in a buffer adjacent to that of the input. The two adjacent buffers can be periodically displayed, and the predicted values optionally mixed with the incoming signal to simulate expectation.

This unit is useful for sensing the metric implications of any k-signal (e.g- the RMS of an audio signal, or the second derivative of a conducting gesture), before sending to a **tempo** statement.

Example:

```
ksum   specsum    wsignal, 1                                         ; sum the amps of a spectrum
ktemp  tempest    ksum, .02, .1, 3, 2, 800, .005, 0, 60, 4, .1, .995  ; and look for beats
```

```
kx, ky    xyin      iprd, ixmin, ixmax, iymin, iymax[, ixinit, iyinit]
          tempo     ktempo, istartempo
```

Sense the cursor position in an input window. Apply tempo control to an uninterpreted score.

INITIALIZATION

iprd - period of cursor sensing (in seconds). Typically .1 seconds.

xmin, *xmax*, *ymin*, *ymax* - edge values for the x-y coordinates of a cursor in the input window.

ixinit, *iyinit* (optional) - initial x-y coordinates reported; the default values are 0,0. If these values are not within the given min-max range, they will be coerced into that range.

istartempo - initial tempo (in beats per minute). Typically 60.

PERFORMANCE

xyin samples the cursor x-y position in an input window every *iprd* seconds. Output values are repeated (not interpolated) at the K-rate, and remain fixed until a new change is registered in the window. There may be any number of input windows. This unit is useful for Realtime control, but continuous motion should be avoided if *iprd* is unusually small.

tempo allows the performance speed of Csound scored events to be controlled from within an orchestra. It operates only in the presence of the **csound -t** flag. When that flag is set, scored events will be performed from their uninterpreted p2 and p3 (beat) parameters, initially at the given command-line tempo. When a **tempo** statement is activated in any instrument (*ktempo* > 0.), the operating tempo will be adjusted to *ktempo* beats per minute. There may be any number of **tempo** statements in an orchestra, but coincident activation is best avoided.

Example:

```
kx,ky   xyin    .05, 30, 0, 120, 0, 75      ; sample the cursor
        tempo   kx, 75                      ; and control the tempo of performance
```

SOUND INPUT & OUTPUT

a1	**in**	
a1, a2	**ins**	
a1, a2, a3, a4	**inq**	
a1	**soundin**	ifilcod[, iskptim][, iformat]
a1, a2	**soundin**	ifilcod[, iskptim][, iformat]
a1, a2, a3, a4	**soundin**	ifilcod[, iskptim][, iformat]
	out	asig
	outs1	asig
	outs2	asig
	outs	asig1, asig2
	outq1	asig
	outq2	asig
	outq3	asig
	outq4	asig
	outq	asig1, asig2, asig3, asig4

These units read/write audio data from/to an external device or stream.

INITIALIZATION

filcod - integer or character-string denoting the source soundfile name. An integer denotes the file **soundin**.*filcod* ; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the named file is sought first in the current directory, then in that given by the environment variable SSDIR (if defined) then by SFDIR. See also **GEN01**.

iskptim (optional) - time in seconds of input sound to be skipped. The default value is 0.

iformat (optional) - specifies the audio data file format (1 = 8-bit signed char, 2 = 8-bit A-law bytes, 3 = 8-bit U-law bytes, 4 = 16-bit short integers, 5 = 32bit long integers, 6 = 32-bit floats). If *iformat* = 0 it is taken from the soundfile header, and if no header from the **csound -o** command flag. The default value is 0.

PERFORMANCE

in, **ins**, **inq** - copy the current values from the standard audio input buffer. If the command-line flag **-i** is set, sound is read continuously from the audio input stream (e.g. *stdin* or a soundfile) into an internal buffer. Any number of these units can read freely from this buffer.

soundin is functionally an audio generator that derives its signal from a pre-existing file. The number of channels read in is set by the number of result cells, a1, a2, etc. A soundin unit opens this file whenever the host instrument is initialized, then closes it again each time the instrument is turned off. There can be any number of **soundin** units within a single instrument or orchestra; also, two or more of them can read simultaneously from the same external file.

out, **outs**, **outq** send audio samples to an accumulating output buffer (created at the beginning of performance) which serves to collect the output of all active instruments before the sound is written to disk. There can be any number of these output units in an instrument. The type (mono, stereo, or quad) must agree with **nchnls**, but units can be chosen to direct sound to any particular channel: **outs1** sends to stereo channel 1, **outq3** to quad channel 3, etc.

a1, a2, a3, a4 **pan** asig, kx, ky, ifn[, imode][, ioffset]

Distribute an audio signal amongst four channels with localization control.

INITIALIZATION

ifn - function table number of a stored pattern describing the amplitude growth in a speaker channel as sound moves towards it from an adjacent speaker. Requires extended guard-point.

imode (optional) - mode of the *kx*, *ky* position values. 0 signifies raw index mode, 1 means the inputs are normalized (0 - 1). The default value is 0.

ioffset (optional) - offset indicator for *kx*, *ky*. 0 infers the origin to be at channel 3 (left rear); 1 requests an axis shift to the quadraphonic center. The default value is 0.

PERFORMANCE

pan takes an input signal *asig* and distributes it amongst four outputs (essentially quad speakers) according to the controls *kx* and *ky*. For normalized input (mode=1) and no offset, the four output locations are in order: left-front at (0,1), right-front at (1,1), left-rear at the origin (0,0), and right-rear at (1,0). In the notation (*kx, ky*), the coordinates *kx* and *ky*, each ranging 0 - 1, thus control the 'rightness' and 'forwardness' of a sound location.

Movement between speakers is by amplitude variation, controlled by the stored function table *ifn*. As *kx* goes from 0 to 1, the strength of the right-hand signals will grow from the left-most table value to the right-most, while that of the left-hand signals will progress from the right-most table value to the left-most. For a simple linear pan, the table might contain the linear function 0 - 1. A more correct pan that maintains constant power would be obtained by storing the first quadrant of a sinusoid. Since pan will scale and truncate *kx* and *ky* in simple table lookup, a medium-large table (say 8193) should be used.

kx, *ky* values are not restricted to 0 - 1. A circular motion passing through all four speakers (enscribed) would have a diameter of root 2, and might be defined by a circle of radius R = root 1/2 with center at (.5,.5). *kx*, *ky* would then come from Rcos(angle), Rsin(angle), with an implicit origin at (.5,.5) (i.e. ioffset = 1). Unscaled raw values operate similarly. Sounds can thus be located anywhere in the polar or cartesian plane; points lying outside the speaker square are projected correctly onto the square's perimeter as for a listener at the center.

Example:

```
            instr   1
     k1     phasor  1/p3                    ; fraction of circle
     k2     tablei  k1, 1, 1                ; sin of angle (sinusoid in f1)
     k3     tablei  k1, 1, 1, .25, 1        ; cos of angle (sin offset 1/4 circle)
     a1     oscili  10000,440, 1            ; audio signal..
a1,a2,a3,a4 pan     a1, k2/2, k3/2, 2, 1, 1 ; sent in a circle (f2=1st quad sin)
            outq    a1, a2, a3, a4
            endin
```

SIGNAL DISPLAY

print	iarg[, iarg,...]
display	xsig, iprd[, iwtflg]
dispfft	xsig, iprd, iwsiz[, iwtyp][, idbouti][, iwtflg]

These units will print orchestra Init-values, or produce graphic display of orchestra control signals and audio signals. Uses X11 windows if enabled, else (or if -g flag is set) displays are approximated in ascii characters.

INITIALIZATION

iprd - the period of display in seconds.

iwsiz - size of the input window in samples. A window of *iwsiz* points will produce a Fourier transform of *iwsiz*/2 points, spread linearly in frequency from 0 to **sr**/2. *iwsiz* must be a power of 2. The windows are permitted to overlap.

iwtyp (optional) - window type. 0 = rectangular, 1 = hanning. The default value is 0 (rectangular).

idbout (optional) - units of output for the Fourier coefficients. 0 = magnitude, 1 = decibels. The default is 0 (magnitude).

iwtflg (optional) - wait flag. If non-zero, each display is held until released by the user. The default value is 0 (no wait).

PERFORMANCE

print - print the current value of the **I**-time arguments (or expressions) *iarg* at every **I**-pass through the instrument.

display - display the audio or control signal *xsig* every *iprd* seconds, as an amplitude vs. time graph.

dispfft - display the Fourier Transform of an audio or control signal (*asig* or *ksig*) every *iprd* seconds using the Fast Fourier Transform method.

Example:

```
k1      envlpx      l, .03, p3, .05, l, .5, .0l      ; generate a note envelope
        display     k1, p3                           ; and display entire shape
```

3. THE STANDARD NUMERIC SCORE

A score is a data file that provides information to an orchestra about its performance. Like an orchestra file, a score file is made up of statements in a known format. The **Csound** orchestra expects to be handed a score comprised mainly of *ascii numeric characters*. Although most users will prefer a higher level score language such as provided by **Cscore**, **Scot**, or another score-generating program, each resulting score must eventually appear in the format expected by the orchestra. A Standard Numeric Score can be created and edited directly by the beginner using standard text editors; indeed, some users continue to prefer it. The purpose of this section is to describe this format in detail.

The basic format of a standard numeric score statement is:

```
opcode p1 p2 p3 p4...          ;comments
```

The *opcode* is a single character, always alphabetic. Legal opcodes are **f, i, a, t, s,** and **e,** the meanings of which are described in the following pages. The opcode is normally the first character of a line; leading spaces or tabs will be ignored. Spaces or tabs between the opcode and *p1* are optional.

p1, p2, p3, etc... are *parameter fields* (*pfields*). Each contains a floating point number comprised of an optional sign, digits, and an optional decimal point. Expressions are not permitted in Standard Score files. *pfields* are separated from each other by one or more spaces or tabs, all but one space of which will be ignored.

Continuation lines are permitted. If the first printing character of a new scoreline is not an opcode, that line will be regarded as a continuation of the pfields from the previous scoreline.

Comments are optional and are for the purpose of permitting the user to document his score file. Comments always begin with a semicolon (;) and extend to the end of the line. Comments will not affect the pfield continuation feature.

Blank lines or comment-only lines are legal (and will be ignored).

Preprocessing of Standard Scores

A Score (a collection of score statements) is divided into time-ordered *sections* by the **s** statement. Before being read by the orchestra, a score is preprocessed one section at a time. Each section is normally processed by 3 routines: Carry, Tempo, and Sort.

1. Carry - within a group of consecutive **i** statements whose p1 whole numbers correspond, any pfield left empty will take its value from the same pfield of the preceding statement. An empty pfield can be denoted by a single point (.) delimited by spaces. No point is required after the last nonempty pfield. The output of Carry preprocessing will show the carried values explicitly. The Carry Feature is not affected by intervening comments or blank lines; it is turned off only by a *non-i* statement or by an **i** statement with unlike p1 whole number.

An additional feature is available for p2 alone. The symbol **+** in p2 will be given the value of p2 + p3 from the preceding **i** statement. This enables note action times to be automatically determined from the sum of preceding durations. The **+** symbol can itself be carried. It is legal only in p2.

E.g.: the statements

```
i1   0    .5   100
i .  +
i
```

will result in

```
i1   0    .5   100
i1   .5   .5   100
i1   1    .5   100
```

The Carry feature should be used liberally. Its use, especially in large scores, can greatly reduce input typing and will simplify later changes.

2. Tempo - this operation time warps a score section according to the information in a **t** statement. The tempo operation converts p2 (and, for **i** statements, p3) from original beats into real seconds, since those are the units required by the orchestra. After time warping, score files will be seen to have orchestra-readable format demonstrated by the following:

i p1 p2beats p2seconds p3beats p3seconds p4 p5 ....

3. Sort - this routine sorts all action-time statements into chronological order by p2 value. It also sorts coincident events into precedence order. Whenever an **f** statement and an **i** statement have the same p2 value, the **f** statement will precede. Whenever two or more **i** statements have the same p2 value, they will be sorted into ascending p1 value order. If they also have the same p1 value, they will be sorted into ascending p3 value order. Score sorting is done section by section (see **s** statement). Automatic sorting implies that score statements may appear in any order within a section.

N.B. The operations Carry, Tempo and Sort are combined in a 3-phase single pass over a score file, to produce a new file in orchestra-readable format (see the Tempo example). Processing can be invoked either explicitly by the **scsort** command, or implicitly by csound which processes the score before calling the orchestra. Source-format files and orchestra-readable files are both in ascii character form, and may be either perused or further modified by standard text editors. Userwritten routines can be used to modify score files before or after the above processes, provided the final orchestra-readable statement format is not violated. Sections of different formats can be sequentially batched; and sections of like format can be merged for automatic sorting.

Next-P and Previous-P Symbols

At the close of any of the above operations, three additional score features are interpreted during file writeout: next-p, previous-p, and ramping.

i statement pfields containing the symbols **np***x* or **pp***x* (where *x* is some integer) will be replaced by the appropriate pfield value found on the next **i** statement (or previous **i** statement) that has the same p1. For example, the symbol np7 will be replaced by the value found in p7 of the next note that is to be played by this instrument. np and pp symbols are recursive and can reference other np and pp symbols which can reference others, etc. References must eventually terminate in a real number or a ramp symbol (see below). Closed loop references should be avoided. np and pp symbols are illegal in p1,p2 and p3 (although they may reference these). np and pp symbols may be Carried. np and pp references cannot cross a Section boundary. Any forward or backward reference to a non-existent note-statement will be given the value zero.

E.g.: the statements

```
i1    0    1    10    np4    pp5
i1    1    1    20
i1    1    1    30
```

will result in

```
i1    0    1    10    20     0
i1    1    1    20    30     20
i1    2    1    30    0      30
```

np and pp symbols can provide an instrument with contextual knowledge of the score, enabling it to glissando or crescendo, for instance, toward the pitch or dynamic of some future event (which may or may not be immediately adjacent). Note that while the Carry feature will propagate np and pp through unsorted statements, the operation that interprets these symbols is acting on a time-warped and fully sorted version of the score.

Ramping

i statement pfields containing the symbol < will be replaced by values derived from linear interpolation of a time-based ramp. Ramps are anchored at each end by the first real number found in the same pfield of a preceding and following note played by the same instrument.

E.g.: the statements

```
i1    0    1    100
i1    1    1    <
i1    2    1    <
i1    3    1    400
i1    4    1    <
i1    5    1    0
```

will result in

```
i1    0    1    100
i1    1    1    200
i1    2    1    300
i1    3    1    400
i1    4    1    200
i1    5    1    0
```

Ramps cannot cross a Section boundary. Ramps cannot be anchored by an np or pp symbol (although they may be referenced by these). Ramp symbols are illegal in p1, p2 and p3. Ramp symbols may be Carried. Note, however, that while the Carry feature will propagate ramp symbols through unsorted statements, the operation that interprets these symbols is acting on a time-warped and fully sorted version of the score. In fact, time-based linear interpolation is based on warped score-time, so that a ramp which spans a group of accelerating notes will remain linear with respect to strict chronological time.

F STATEMENT (or FUNCTION TABLE STATEMENT)

f p1 p2 p3 p4 ...

This causes a **GEN** subroutine to place values in a stored function table for use by instruments.

PFIELDS

p1	Table number (from 1 to 200) by which the stored function will be known. A negative number requests that the table be destroyed.
p2	Action time of function generation (or destruction) in beats.
p3	Size of function table (i.e. number of points). Must be a power of 2, or a power-of-2 plus 1 (see below). Maximum table size is 16777216 (2**24) points.
p4	Number of the GEN routine to be called (see GEN ROUTINES). A negative value will cause rescaling to be omitted.
p5 p6 . .	Parameters whose meaning is determined by the particular GEN routine.

SPECIAL CONSIDERATIONS

Function tables are arrays of floating-point values. Arrays can be of any length in powers of 2; space allocation always provides for 2**n points plus an additional *guard point*. The guard point value, used during interpolated lookup, can be automatically set to reflect the table's purpose: If *size* is an exact power of 2, the guard point will be a copy of the first point; this is appropriate for *interpolated wrap-around* lookup as in **oscili**, etc., and should even be used for non-interpolating **oscil** for safe consistency. If *size* is set to 2**n + 1, the guard point value automatically extends the contour of table values; this is appropriate for single-scan functions such in **envlpx**, **oscil1**, **oscil1i**, etc.

Table space is allocated in primary memory, along with instrument data space. The maximum table number has a soft limit of 200; this can be extended if required.

An existing function table can be removed by an **f** statement containing a negative p1 and an appropriate action time. A function table can also be removed by the generation of another table with the same p1. Functions are not automatically erased at the end of a score section.

p2 action time is treated in the same way as in **i** statements with respect to sorting and modification by **t** statements. If an **f** statement and an **i** statement have the same p2, the sorter gives the **f** statement precedence so that the function table will be available during note initialization.

An **f 0** statement (zero p1, positive p2) may be used to create an action time with no associated action. Such time markers are useful for padding out a score section (see **s** statement).

I STATEMENT (INSTRUMENT or NOTE STATEMENT)

i p1 p2 p3 p4 ...

This statement calls for an instrument to be made active at a specific time and for a certain duration. The parameter field values are passed to that instrument prior to its initialization, and remain valid throughout its Performance.

PFIELDS

p1 Instrument number (from 1 to 200), usually a non-negative integer. An optional fractional part can provide an additional tag for specifying ties between particular notes of consecutive clusters. A negative p1 (including tag) can be used to turn off a particular 'held' note.

p2 Starting time in arbitrary units called beats.

p3 Duration time in beats (usually positive). A negative value will initiate a held note (see also **ihold**). A zero value will invoke an initialization pass without performance (see also **instr**).

p4
p5 Parameters whose significance is determined by the instrument.
.
.

SPECIAL CONSIDERATIONS

Beats are evaluated as seconds, unless there is a **t** statement in this score section or a **-t** flag in the command line.

Starting or action times are relative to the beginning of a section (see **s** statement), which is assigned time 0.

Note statements within a section may be placed in any order. Before being sent to an orchestra, unordered score statements must first be processed by Sorter, which will reorder them by ascending p2 value. Notes with the same p2 value will be ordered by ascending p1; if the same p1, then by ascending p3.

Notes may be stacked, i.e., a single instrument can perform any number of notes simultaneously. (The necessary copies of the instrument's data space will be allocated dynamically by the orchestra loader.) Each note will normally turn off when its p3 duration has expired, or on receipt of a MIDI noteoff signal. An instrument can modify its own duration either by changing its p3 value during note initialization, or by prolonging itself through the action of a **linenr** unit.

An instrument may be turned on and left to perform indefinitely either by giving it a negative p3 or by including an **ihold** in its **I**-time code. If a held note is active, an **i** statement *with matching p1* will not cause a new allocation but will take over the data space of the held note. The new pfields (including p3) will now be in effect, and an **I**-time pass will be executed in which the units can either be newly initialized or allowed to continue as required for a tied note (see **tigoto**). A held note may be succeeded either by another held note or by a note of finite duration. A held note will continue to perform across section endings (see **s** statement). It is halted only by **turnoff** or by an **i** statement with negative matching p1 or by an **e** statement.

A STATEMENT (or ADVANCE STATEMENT)

a p1 p2 p3

This causes score time to be advanced by a specified amount without producing sound samples.

PFIELDS

p1	carries no meaning. Usually zero
p2	Action time, in beats, at which advance is to begin.
p3	Durational span (distance in beats) of time advance.

p4	
p5	These carry no meaning.
.	
.	

SPECIAL CONSIDERATIONS

This statement allows the beat count within a score section to be advanced without generating intervening sound samples. This can be of use when a score section is incomplete (the beginning or middle is missing) and the user does not wish to generate and listen to a lot of silence.

p2 action time and p3 distance are treated as in **i** statements, with respect to sorting and modification by **t** statements.

An **a** statement will be temporarily inserted in the score by the Score Extract feature when the extracted segment begins later than the start of a Section. The purpose of this is to preserve the beat count and time count of the original score for the benefit of the *peak amplitude messages* which are reported on the user console.

Whenever an **a** statement is encountered by a performing orchestra, its presence and effect will be reported on the user's console.

T STATEMENT (TEMPO STATEMENT)

t p1 p2 p3 p4 ..... (unlimited)

This statement sets the tempo and specifies the accelerations and decelerations for the current section. This is done by converting beats into seconds.

PFIELDS

p1	must be zero
p2	initial tempo in beats per minute
p3, p5, p7, ...	times in beats (in non-decreasing order)
p4, p6, p8, ...	tempi for the referenced beat times

SPECIAL CONSIDERATIONS

Time and Tempo-for-that-time are given as ordered couples that define points on a "tempo vs time" graph. (The time-axis here is in beats so is not necessarily linear.) The beat-rate of a Section can be thought of as a movement from point to point on that graph: motion between two points of equal height signifies constant tempo, while motion between two points of unequal height will cause an accelarando or ritardando accordingly. The graph can contain discontinuities: two points given equal times but different tempi will cause an immediate tempo change.

Motion between different tempos over non-zero time is inverse linear. That is, an accelerando between two tempos M1 and M2 proceeds by linear interpolation of the single-beat durations from 60/M1 to 60/M2.

The first tempo given must be for beat 0.

A tempo, once assigned, will remain in effect from that time-point unless influenced by a succeeding tempo, i.e. the last specified tempo will be held to the end of the section.

A **t** statement applies only to the score section in which it appears. Only one **t** statement is meaningful in a section; it can be placed anywhere within that section. If a score section contains no **t** statement, then beats are interpreted as seconds (i.e. with an implicit **t 0 60** statement).

N.B. If the **csound** command includes a **-t** flag, the interpreted tempo of all score **t** statements will be overridden by the command-line tempo.

S STATEMENT

s anything

The **s** statement marks the end of a section.

PFIELDS

All pfields are ignored.

SPECIAL CONSIDERATIONS

Sorting of the **i**, **f** and **a** statements by action time is done section by section.

Time warping for the **t** statement is done section by section.

All action times within a section are relative to its beginning. A section statement establishes a new relative time of 0, but has no other reinitializing effects (e.g. stored function tables are preserved across section boundaries).

A section is considered complete when all action times and finite durations have been satisfied (i.e., the "length" of a section is determined by the last occurring action or turn-off). A section can be extended by the use of an **f 0** statement.

A section ending automatically invokes a Purge of inactive instrument and data spaces.

N.B. Since score statements are processed section by section, the amount of memory required depends on the maximum number of score statements in a section. Memory allocation is dynamic, and the user will be informed as extra memory blocks are requested during score processing.

For the end of the final section of a score, the **s** statement is optional; the **e** statement may be used instead.

E STATEMENT

e anything

This statement may be used to mark the end of the last section of the score.

PFIELDS

All pfields are ignored.

SPECIAL CONSIDERATIONS

The **e** statement is contextually identical to an **s** statement. Additionally, the **e** statement terminates all signal generation (including indefinite performance) and closes all input and output files.

If an **e** statement occurs before the end of a score, all subsequent score lines will be ignored.

The **e** statement is optional in a score file yet to be sorted. If a score file has no **e** statement, then Sort processing will supply one.

4. GEN ROUTINES

The GEN subroutines are function-drawing procedures called by **f** statements to construct stored wavetables. They are available throughout orchestra performance, and can be invoked at any point in the score as given by p2. p1 assigns a table *number*, and p3 the table *size* (see **f** statement). p4 specifies the GEN routine to be called; each GEN routine will assign special meaning to the pfield values that follow.

GEN01

This subroutine transfers data from a soundfile into a function table.

f # time size 1 filcod skiptime format

size - number of points in the table. Ordinarily a power of 2 or a power-of-2 plus 1 (see **f** statement); the maximum tablesize is 16777216 (2**24) points. If the source soundfile is of type AIFF, allocation of table memory can be *deferred* by setting this parameter to 0; the size allocated is then the number of points in the file (probably not a power-of-2), and the table is not usable by normal oscillators, but it is usable by a **loscil** unit. An AIFF source can also be mono or stereo.

filcod - integer or character-string denoting the source soundfile name. An integer denotes the file **soundin**.*filcod* ; a character-string (in double quotes, spaces permitted) gives the filename itself, optionally a full pathname. If not a full path, the file is sought first in the current directory, then in that given by the environment variable SSDIR (if defined) then by SFDIR. See also **soundin**.

skiptime - begin reading at *skiptime* seconds into the file.

format - specifies the audio data-file format:

1 - 8-bit signed character	4 - 16-bit short integers
2 - 8-bit A-law bytes	5 - 32-bit long integers
3 - 8-bit U-law bytes	6 - 32-bit floats

If *format* = 0 the sample format is taken from the soundfile header, or by default from the **csound -o** command flag.

Reading stops at end-of-file or when the table is full. Table locations not filled will contain zeros.

Note:

If p4 is positive, the table will be post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative p4 will cause rescaling to be skipped.

Examples:

```
f  1  0  8192  1  23  0  4
f  2  0  0  -1  "trumpet  A#5"  0  4
```

The tables are filled from 2 files, “soundin.23” and “trumpet A#5”, expected in SSDIR or SFDIR. The first table is pre-allocated; the second is allocated dynamically, and its rescaling is inhibited.

GEN02

This subroutine transfers data from immediate pfields into a function table.

f # time size 2 v1 v2 v3 . . .

size - number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see **f** statement). The maximum tablesize is 16777216 (2**24) points.

v1, v2, v3, ... - values to be copied directly into the table space. The number of values is limited by the compile-time variable PMAX, which controls the maximum pfields (currently 150). The values copied may include the table guard point; any table locations not filled will contain zeros.

Note:

If p4 is positive, the table will be post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative p4 will cause rescaling to be skipped.

Example:

```
f 1 0 16 -2 0 1 2 3 4 5 6 7 8 9 10 11 0
```

This calls upon **GEN02** to place 12 values plus an explicit wrap-around guard value into a table of size next-highest power of 2. Rescaling is inhibited.

GEN03

This subroutine generates a stored function table by evaluating a polynomial in x over a fixed interval and with specified coefficients.

f # time size 3 xval1 xval2 c0 c1 c2 . . . cn

size - number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see **f** statement).

xval1, xval2 - left and right values of the x interval over which the polynomial is defined (*xval1* < *xval2*). These will produce the 1st stored value and the (power-of-2 plus l)th stored value respectively in the generated function table.

c0, c1, c2, ... cn - coefficients of the nth-order polynomial

$$c_0 + c_1x + c_2x^2 + \ldots + c_nx^n$$

Coefficients may be positive or negative real numbers; a zero denotes a missing term in the polynomial. The coefficient list begins in p7, providing a current upper limit of 144 terms.

Note:

The defined segment [fn(*xval1*),fn(*xval2*)] is evenly distributed. Thus a 512-point table over the interval [-1,1] will have its origin at location 257 (at the start of the 2nd half). Provided the extended guard point is requested, both fn(-1) and fn(1) will exist in the table.

GEN03 is useful in conjunction with **table** or **tablei** for audio waveshaping (sound modification by non-linear distortion). Coefficients to produce a particular formant from a sinusoidal lookup index of known amplitude can be determined at preprocessing time using algorithms such as Chebyshev formulae. See also **GEN13**.

Example:

```
f 1 0 1025 3 -1 1 5 4 3 2 2 1
```

This calls **GEN03** to fill a table with a 4th order polynomial function over the x-interval -1 to 1. The origin will be at the offset position 512. The function is post-normalized.

GEN04

This subroutine generates a normalizing function by examining the contents of an existing table.

f # time size 4 source# sourcemode

size - number of points in the table. Should be power-of-2 plus 1. Must not exceed (except by 1) the size of the source table being examined; limited to just half that size if the sourcemode is of type offset (see below).

source # - table number of stored function to be examined.

sourcemode - a coded value, specifying how the source table is to be scanned to obtain the normalizing function. Zero indicates that the source is to be scanned from left to right. Non-zero indicates that the source has a bipolar structure; scanning will begin at the mid-point and progress outwards, looking at pairs of points equidistant from the center.

Note:

The normalizing function derives from the progressive absolute maxima of the source table being scanned. The new table is created left-to-right, with stored values equal to 1/(absolute maximum so far scanned). Stored values will thus begin with 1/(first value scanned), then get progressively smaller as new maxima are encountered. For a source table which is normalized (values <= 1), the derived values will range from 1/(first value scanned) down to 1. If the first value scanned is zero, that inverse will be set to 1.

The normalizing function from **GEN04** is not itself normalized.

GEN04 is useful for scaling a table-derived signal so that it has a consistent peak amplitude. A particular application occurs in waveshaping when the carrier (or indexing) signal is less than full amplitude.

Example:

```
f  2  0  512  4  1  1
```

This creates a normalizing function for use in connection with the **GEN03** table 1 example. Midpoint bipolar offset is specified.

GEN05, GEN07

These subroutines are used to construct functions from segments of exponential curves (**GEN05**) or straight lines (**GEN07**).

f #   time   size   5   a   n1   b   n2   c . . .
f #   time   size   7   a   n1   b   n2   c . . .

size - number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see **f** statement).

a, b, c, etc. - ordinate values, in odd-numbered pfields p5, p7, p9, . . . For **GEN05** these must be nonzero and must be alike in sign. No such restrictions exist for **GEN07**.

n1, n2, etc. - length of segment (no. of storage locations), in even-numbered pfields. Cannot be negative, but a zero is meaningful for specifying discontinuous waveforms (e.g. in the example below). The sum n1 + n2 + ... will normally equal *size* for fully specified functions. If the sum is smaller, the function locations not included will be set to zero; if the sum is greater, only the first *size* locations will be stored.

Note:

If p4 is positive, functions are post-normalized (rescaled to a maximum absolute value of 1 after generation). A negative p4 will cause rescaling to be skipped.

Discrete-point linear interpolation implies an increase or decrease along a segment by equal differences between adjacent locations; exponential interpolation implies that the progression is by equal ratio. In both forms the interpolation from *a* to *b* is such as to assume that the value *b* will be attained in the n + 1th location. For discontinuous functions, and for the segment encompassing the end location, this value will not actually be reached, although it may eventually appear as a result of final scaling.

Example:

```
f  1  0  256  7  0  128  1  0  -1  128  0
```

This describes a single-cycle sawtooth whose discontinuity is mid-way in the stored function.

GEN06

This subroutine will generate a function comprised of segments of cubic polynomials, spanning specified points just three at a time.

f # time size 6 a n1 b n2 c n3 d . . .

size - number of points in the table. Must be a power off or power-of-2 plus 1 (see **f** statement).

a, c, e, ... - local maxima or minima of successive segments, depending on the relation of these points to adjacent inflexions. May be either positive or negative.

b, d, f, ... - ordinate values of points of inflexion at the ends of successive curved segments. May be positive or negative.

n1, n2, n3... - number of stored values between specified points. Cannot be negative, but a zero is meaningful for specifying discontinuities. The sum n1 + n2 + ... will normally equal *size* for fully specified functions. (for details, see **GEN05**).

Note:

GEN06 constructs a stored function from segments of cubic polynomial functions. Segments link ordinate values in groups of 3: point of inflexion, maximum/minimum, point of inflexion. The first complete segment encompasses *b,c,d* and has length *n2 + n3*, the next encompasses *d,e,f* and has length *n4 + n5*, etc. The first segment (*a,b* with length *n1*) is partial with only one inflexion; the last segment may be partial too. Although the inflexion points *b,d,f* ... each figure in two segments (to the left and right), the slope of the two segments remains independent at that common point (i.e. the 1st derivative will likely be discontinuous). When *a,c,e*... are alternately maximum and minimum, the inflexion joins will be relatively smooth; for successive maxima or successive minima the inflexions will be comb-like.

Example:

```
f   1   0   65   6   0   16   .5   16   1   16   0   16   -1
```

This creates a curve running 0 to 1 to -1, with a minimum, maximum and minimum at these values respectively. Inflexions are at .5 and 0, and are relatively smooth.

GEN08

This subroutine will generate a piecewise cubic spline curve, the smoothest possible through all specified points.

f # time size 8 a n1 b n2 c n3 d . . .

size - number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see **f** statement).

a, b, c ... - ordinate values of the function.

n1, n2, n3 ... - length of each segment measured in stored values. May not be zero, but may be fractional. A particular segment may or may not actually store any values; stored values will be generated at integral points from the beginning of the function. The sum n1 + n2 + ... will normally equal *size* for fully specified functions.

Note:

GEN08 constructs a stored table from segments of cubic polynomial functions. Each segment runs between two specified points but depends as well on their neighbors on each side. Neighboring segments will agree in both value and slope at their common point. (The common slope is that of a parabola through that point and its two neighbors). The slope at the two ends of the function is constrained to be zero (flat).

Hint: to make a discontinuity in slope or value in the function as stored, arrange a series of points in the interval between two stored values; likewise for a non-zero boundary slope.

Examples:

f 1 0 65 8 0 16 0 16 1 16 0 16 0

This example creates a curve with a smooth hump in the middle, going briefly negative outside the hump then flat at its ends.

f 2 0 65 8 0 16 0 .1 0 15.9 1 15.9 0 .1 0 16 0

This example is similar, but does not go negative.

GEN09, GEN10, GEN19

These subroutines generate composite waveforms made up of weighted sums of simple sinusoids. The specification of each contributing partial requires 3 pfields using **GEN09,** 1 using **GEN10,** and 4 using **GEN19.**

f	#	time	size	9	pna	stra	phsa	pnb	strb	phsb . . .		
f	#	time	size	10	str1	str2	str3	str4				
f	#	time	size	19	pna	stra	phsa	dcoa	pnb	strb	phsb	dcob . . .

size - number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see **f** statement).

pna, pnb, etc. - partial no. (relative to a fundamental that would occupy *size* locations per cycle) of sinusoid a, sinusoid b, etc. Must be positive, but need not be a whole number, i.e., non-harmonic partials are permitted. Partials may be in any order.

stra, strb, etc. - strength of partials *pna, pnb,* etc. These are relative strengths, since the composite waveform may be rescaled later. Negative values are permitted and imply a 180 degree phase shift.

phsa, phsb, etc. - initial phase of partials *pna, pnb*, etc., expressed in degrees.

dcoa, dcob, etc. - DC offset of partials *pna, pnb*, etc. This is applied *after* strength scaling, i.e. a value of 2 will lift a 2-strength sinusoid from range [-2,2] to range [0,4] (before later rescaling).

str1, str2, str3, etc. - relative strengths of the fixed harmonic partial numbers 1,2,3, etc., beginning in p5. Partials not required should be given a strength of zero.

Note:

These subroutines generate stored functions as sums of sinusoids of different frequencies. The two major restrictions on **GEN10**—that the partials be harmonic and in phase—do not apply to **GEN09** or **GEN19**.

In each case the composite wave, once drawn, is then rescaled to unity if p4 was positive. A negative p4 will cause rescaling to be skipped.

Examples:

```
f  1  0  1024   9   1  3    0  3  1  0  9  .3333  180
f  2  0  1024  19  .5  1  270  1
```

f 1 combines partials l, 3 and 9 in the relative strengths in which they are found in a square wave, except that partial 9 is upside down. f 2 creates a rising sigmoid [0 - 2]. Both will be rescaled.

GEN11

This subroutine generates an additive set of cosine partials, in the manner of Csound generators **buzz** and **gbuzz**.

f # time size 11 nh lh r

size - number of points in the table. Must be a power of 2 or power-of-2 plus 1 (see **f** statement).

nh - number of harmonics requested. Must be positive.

lh (optional) - lowest harmonic partial present. Can be positive, zero or negative. The set of partials can begin at any partial number and proceeds upwards; if *lh* is negative, all partials below zero will reflect in zero to produce positive partials without phase change (since cosine is an even function), and will add constructively to any positive partials in the set. The default value is 1.

r (optional) - multiplier in an amplitude coefficient series. This is a power series: if the *lh*th partial has a strength coefficient of **A** the (*lh* + n)th partial will have a coefficient of **A** * *r***n, i.e. strength values trace an exponential curve. *r* may be positive, zero or negative, and is not restricted to integers. The default value is 1.

Note:

This subroutine is a non-time-varying version of the csound **buzz** and **gbuzz** generators, and is similarly useful as a complex sound source in subtractive synthesis. With *lh* and *r* present it parallels **gbuzz**; with both absent or equal to 1 it reduces to the simpler **buzz** (i.e. *nh* equal-strength harmonic partials beginning with the fundamental).

Sampling the stored waveform with an oscillator is more efficient than using dynamic buzz units. However, the spectral content is invariant, and care is necessary lest the higher partials exceed the Nyquist during sampling to produce foldover.

Examples:

```
f 1   0   2049   11   4
f 2   0   2049   11   4   1   1
f 3   0   2049  -11   7   3  .5
```

The first two tables will contain identical band-limited pulse waves of four equal-strength harmonic partials beginning with the fundamental. The third table will sum seven consecutive harmonics, beginning with the third, and at progressively weaker strengths (1, .5, .25, .125 . . .). It will not be post-normalized.

GEN12

This generates the log of a modified Bessel function of the second kind, order 0, suitable for use in amplitude-modulated FM.

f # time size -12 xint

size - number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see **f** statement). The normal value is power-of-2 plus 1.

xint - specifies the **x** interval [*0 to +int*] over which the function is defined.

Note:

This subroutine draws the natural log of a modified Bessel function of the second kind, order 0 (commonly written as **I** subscript 0), over the x-interval requested. The call should have rescaling inhibited.

The function is useful as an amplitude scaling factor in cycle-synchronous amplitude-modulated FM. (See Palamin & Palamin, *J. Audio Eng. Soc., 36/9*, Sept. 1988, pp.671-684.) The algorithm is interesting because it permits the normally symmetric FM spectrum to be made asymmetric around a frequency other than the carrier, and is thereby useful for formant positioning. By using a table lookup index of **I**(*r - 1/r*), where **I** is the FM modulation index and **r** is an exponential parameter affecting partial strengths, the Palamin algorithm becomes relatively efficient, requiring only oscil's, table lookups, and a single *exp* call.

Example:

f 1 0 2049 -12 20

This draws an unscaled *ln(I0(x))* from 0 to 20.

GEN13, GEN14

These subroutines use Chebyshev coefficients to generate stored polynomial functions which, under waveshaping, can be used to split a sinusoid into harmonic partials having a predefinable spectrum.

```
f  #  time  size  13  xint  xamp  h0  h1  h2 . . . hn
f  #  time  size  14  xint  xamp  h0  h1  h2 . . . hn
```

size - number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see **f** statement). The normal value is power-of-2 plus 1.

xint - provides the left and right values [*-xint*, *+xint*] of the *x* interval over which the polynomial is to be drawn. These subroutines both call **GEN03** to draw their functions; the p5 value here is therefor expanded to a negative-positive p5,p6 pair before **GEN03** is actually called. The normal value is 1.

xamp - amplitude scaling factor of the sinusoid input that is expected to produce the following spectrum.

h0, h1, h2, ... hn - relative strength of partials 0 (DC), 1 (fundamental), 2 ... that will result when a sinusoid of amplitude *xamp * int(size/2)/xint* is waveshaped using this function table. These values thus describe a frequency spectrum associated with a particular factor *xamp* of the input signal.

Note:

GEN13 is the function generator normally employed in standard waveshaping. It stores a polynomial whose coefficients derive from the Chebyshev polynomials of the first kind, so that a driving sinusoid of strength *xamp* will exhibit the specified spectrum at output. Note that the evolution of this spectrum is generally not linear with varying *xamp*. However, it is bandlimited (the only partials to appear will be those specified at generation time); and the partials will tend to occur and to develop in ascending order (the lower partials dominating at low *xamp*, and the spectral richness increasing for higher values of *xamp*). A negative *hn* value implies a 180 degree phase shift of that partial; the requested full-amplitude spectrum will not be affected by this shift, although the evolution of several of its component partials may be. The pattern +,+,-,-,+,+,... for *h0,h1,h2...* will minimize the normalization problem for low *xamp* values (see above), but does not necessarily provide the smoothest pattern of evolution.

GEN14 stores a polynomial whose coefficients derive from Chebyshevs of the second kind.

Example:

```
f  1  0  1025  13  1  1  0  5  0  3  0  1
```

This creates a function which, under waveshaping, will split a sinusoid into 3 odd-harmonic partials of relative strength 5:3:1.

GEN15

This subroutine creates two tables of stored polynomial functions, suitable for use in phase quadrature operations.

f # time size 15 xint xamp h0 phs0 h1 phs1 h2 phs2 . . .

size - number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see **f** statement). The normal value is power-of-2 plus 1.

xint - provides the left and right values [-*xint*, +*xint*] of the *x* interval over which the polynomial is to be drawn. This subroutine will eventually call **GEN03** to draw both functions; this p5 value is therefor expanded to a negative-positive p5, p6 pair before **GEN03** is actually called. The normal value is 1.

xamp - amplitude scaling factor of the sinusoid input that is expected to produce the following spectrum.

h0, h1, h2, ... hn - relative strength of partials 0 (DC), 1 (fundamental), 2 ... that will result when a sinusoid of amplitude *xamp * int(size/2)/xint* is waveshaped using this function table. These values thus describe a frequency spectrum associated with a particular factor *xamp* of the input signal.

phs0, phs1, ... - phase in degrees of desired harmonics *h0, h1, ...* when the two functions of **GEN15** are used with phase quadrature.

Note:

GEN15 creates two tables of equal size, labelled **f** # and **f** # + 1. Table # will contain a Chebyshev function of the first kind, drawn using **GEN03** with partial strengths *h0cos*(*phs0*), *h1cos*(*phs1*), ... Table #+*1* will contain a Chebyshev function of the 2nd kind by calling **GEN14** with partials *h1sin*(*phs1*), *h2sin*(*phs2*),... (note the harmonic displacement). The two tables can be used in conjunction in a waveshaping network that exploits phase quadrature.

GEN17

This subroutine creates a step function from given x-y pairs.

f # time size 17 x1 a x2 b x3 c . . .

size - number of points in the table. Must be a power of 2 or a power-of-2 plus 1 (see **f** statement). The normal value is power-of-2 plus 1.

x1, x2, x3, etc. - x-ordinate values, in ascending order, 0 first.

a, b, c, etc. - y-values at those x-ordinates, held until the next x-ordinate.

This subroutine creates a step function of x-y pairs whose y-values are held to the right. The right-most y-value is then held to the end of the table. The function is useful for mapping one set of data values onto another, such as MIDI note numbers onto sampled sound ftable numbers (see **loscil**).

Example:

```
f 1 0 128 -17 0 1 12 2 24 3 36 4 48 5 60 6 72 7 84 8
```

This describes a step function with 8 successively increasing levels, each 12 locations wide except the last which extends its value to the end of the table. Rescaling is inhibited. Indexing into this table with a MIDI note-number would retrieve a different value every octave up to the eighth, above which the value returned would remain the same.

5. SCOT: A Score Translator

Scot is a language for describing scores in a fashion that parallels traditional music notation. **Scot** is also the name of a program which translates scores written in this language into *standard numeric score* format so that the score can be performed by **Csound**. The result of this translation is placed in a file called *score*. A score file written in Scot (named *file.sc*, say) can be sent through the translator by the command

```
scot  file.sc
```

The resulting numeric score can then be examined for errors, edited, or performed by typing

```
csound  file.orc  score
```

Alternatively, the command

```
csound  file.orc  -S file.sc
```

would combine both processes by informing **Csound** of the initial score format.

Internally, a Scot score has at least three parts: a section to define instrument names, a section to define functions, and one or more actual score sections. It is generally advisable to keep score sections short to facilitate finding errors. The overall layout of a Scot score has three main sections:

```
orchestra { .... }
functions { .... }
score { .... }
```

The last two sections may be repeated as many times as desired. The functions section is also optional. Throughout this **Scot** document, bear in mind that you are free to break up each of these divisions into as many lines as seem convenient, or to place a carriage return anywhere you are allowed to insert a space, including before and after the curly brackets. Furthermore, you may use as many spaces or tabs as you need to make the score easy to read. **Scot** imposes no formatting restrictions except that numbers, instrument names, and keywords (for example, *orchestra*) may not be broken with spaces. You may insert comments (such as measure numbers) anywhere in the score by preceding them with a semicolon. A semicolon causes **Scot** to ignore the rest of a line.

Orchestra Declaration Section

The orchestra section of a **Scot** score serves to designate instrument names for use within the score. This is a matter of convenience, since an orchestra knows instruments only by numbers, not names. If you declare three instruments, such as:

```
orchestra { flute=1 cello=2 trumpet=3 }
```

Csound will neither know nor care what you have named the note lists. However, when you use the name *$flute,* **Scot** will know you are referring to **instr 1** in the orchestra, *$cello* will refer to **instr 2**, and *$trumpet* will be **instr 3**. You may meaningfully skip numbers or give several instruments the same number. It is up to you to make sure that your orchestra has the correct instruments and that the association between these names and the instruments is what you intend. There is no limit (or a very high one, at least) as to how many instruments you can declare.

Function Declaration Section

The major purpose of this division is to allow you to declare function tables for waveforms, envelopes, etc. These functions are declared exactly as specified for **Csound**. In fact, everything you type between the brackets in this section will be passed directly to the resulting *numeric* score with no modification, so that mistakes will not be caught by the **Scot** program, but rather by the subsequent performance. You can use this section to write notes for instruments for which traditional pitch-rhythm notation is inappropriate. The most common example would be turning on a reverb instrument. Instruments referenced in this way need not appear in the **Scot** orchestra declaration. Here is a possible function declaration:

```
functions {
f1 0 256 10 1 0 .5 0 .3
f2 0 256 7 0 64 1 64 .7 64 0
i9 0 -1 3                    ; this turns on instr 9
}
```

Score Section

The **Scot** statements contained inside the braces of each score statement is translated into a numeric score Section (q.v.). It is wise to keep score sections small, say seven or eight measures of five voices at a time. This avoids overloading the system, and simplifies error checking.

The beginning of the score section is specified by typing:

```
score {
```

Everything which follows until the braces are closed is within a single section. Within this section you write measures of notes in traditional pitch and rhythm notation for any of the instrument names listed in your orchestra declaration. These notes may carry additional information such as slurs, ties and parameter fields. Let us now consider the format for notes entered in a **Scot** score.

The first thing to do is name the instrument you want and the desired meter. For example, to write some 4/4 measures for the cello, type:

```
$cello
!ti "4/4"
```

The dollar sign and exclamation point tell Scot that a special declarator follows. The time signature declarator is optional; if present, **Scot** will check the number of beats in each measure for you.

Pitch and Rhythm

The two basic components of a note statement are the pitch and duration. Pitch is specified using the alphabetic note name, and duration is specified using numeric characters. Duration is indicated at the beginning of the note as a number representing the division of a whole beat. You may always find the number specifying a given duration by thinking of how many times that duration would fit in a 4/4 measure. Also, if the duration is followed by a dot (`.') it is increased by 50%, exactly as in traditional notation. Some sample durations are listed below:

whole note	1
half note	2
double dotted quarter	4..

dotted quarter note	4.
quarter note	4
half note triplet	6
eighth note	8
eighth note triplet	12
sixteenth note	16
thirty-second note	32

Pitch is indicated next by first (optionally) specifying the register and then the note name, followed by an accidental if desired. Normally, the "octave following" feature is in effect. This feature causes any note named to lie within the interval of an augmented fourth of the previous note, unless a new register is chosen. The first note you write will always be within a fourth of middle c unless you choose a different register.

For example, if the first note of an instrument part is notated g flat, the scot program assigns the pitch corresponding to the g flat below middle c. On the other hand, if the first note is f sharp, the pitch assigned will be the f sharp above middle c. Changes of register are indicated by a preceding apostrophe for each octave displacement upward or a preceding comma for each octave displacement downward. Commas and apostrophes always displace the pitch by the desired number of octaves starting from that note which is within an augmented fourth of the previous pitch.

If you ever get lost, prefacing the pitch specification with an `=' returns the reference to middle c. It is usually wise to use the equals sign in your first note statement and whenever you feel uncertain as to what the current registration is. Let us now write two measures for the cello part, the first starting in the octave below middle c and the second repeating but starting in the octave above middle c:

```
$cello
!ti "4/4"
4=g 4e 4d 4c/ 4='g 4e 4d 4c
```

As you can see, a slash indicates a new measure and we have chosen to use the dummy middle c to indicate the new register. A more convenient way of notating these two measures would be to type the following:

```
$cello
!ti "4/4"
4=g e d c/ ''g e d c
```

You may observe in this example that the quarter note duration carries to the following notes when the following durations are left unspecified. Also, two apostrophes indicate an upward pitch displacement of two octaves from two g's below middle c, where the pitch would have fallen without any modification. It is important to remember three things, then, when specifying pitches:

1) Note pitches specified by letter name only (with or without accidental) will always fall within an interval of a fourth from the preceding pitch.

2) These pitches can be octave displaced upward or downward by preceding the note letter with the desired number of apostrophes or commas.

3) If you are unsure of the current register, you may begin the pitch component of the note with an equals sign which acts as a dummy middle c.

The pitch may be modified by an accidental after the note name:

n	natural
#	sharp
- (hyphen)	flat
##	double sharp
-- (double hyphen)	double flat

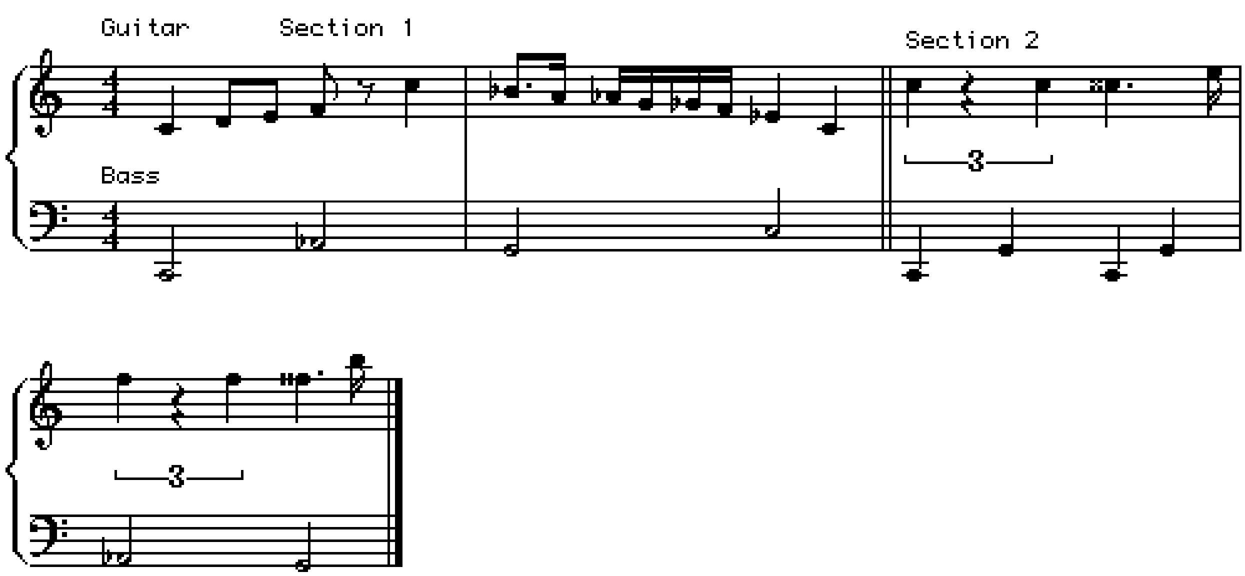
Accidentals are carried throughout the measure just as in traditional music notation. However, an accidental specified within a measure will hold for that note in all registers, in contrast with traditional notation. Therefore, make sure to specify *n* when you no longer want an accidental applied to that pitch-class.

Pitches entered in the Scot score are translated into the appropriate octave point pitch-class value and appear as parameter p5 in the numeric score output. This means you must design your instruments to accept p5 as pitch.

Rests are notated just like notes but using the letter *r* instead of a pitch name. *4r* therefore indicates a quarter rest and *1r* a whole rest. Durations carry from rest to rest or to following pitches as mentioned above.

The tempo in beats per minute is specified in each section by choosing a single instrument part and using tempo statements (e.g. t90) at the various points in the score as needed. A quarter note is interpreted as a single beat, and tempi are interpolated between the intervening beats (see score **t** statement).

Scot Example I



A Scot encoding of this score might appear as follows:

```
; A BASIC Tune
orchestra        { guitar=1 bass=2 }
functions  {
f1 0 512 10 1 .5 .25 .126
f2 0 256  7 1 120 1 8 0 128 1
}
score    {  ;section 1
$guitar
!ti "4/4"
4=c 8d e- f r 4='c/
8.b- 16a a- g  g- f  4e- c/
$bass
2=,,c  'a-/
g  =,c/
}
score    {  ;section 2
$guitar
!ti "4/4"
6='c r c 4..c## 16e- /
6f r f 4..f## 16b /
$bass
4=,,c  'g,  c  'g/
2=a-  g /
}
```

The score resulting from this Scot notation is shown at the end of this chapter.

Groupettes

Duration numbers can have any integral value; for instance,

```
!time "4/4"
5cdefg/
```

would encode a measure of *5 in the time of 4* quarter notes. However, specification of arbitrary rhythmic groupings in this way is at best awkward. Instead, arbitrary portions of the score section may be enclosed in *groupette brackets*. The durations of all notes inside groupette brackets will be multiplied by a fraction *n/d*, where the musical meaning is *d* in the time of *n*. Assuming *d* and *n* here are integers, groupette brackets may take these several forms:

{d:n:	....	:}	d in the time of n
{d::	....	:}	n will be the largest power of 2 less than d
{:	....	:}	d=3, n=2 (normal triplets)

It can be seen that the second and third form are abbreviations for the more common kinds of groupettes. (Observe the punctuation of each form carefully.) Groupettes may be nested to a reasonable depth. Also, groupette factors apply only after the written duration is carried from note to note. Thus, the following example is a correct specification for two measures of 6/8 time:

```
!time "6/8" 8cde {4:3: fgab :} / crc 4.c /
```

The notes inside the groupette are *4 in the space of 3* 8th notes, and the written-8th-note duration carries normally into the next measure. This closely parallels the way groupette brackets and note durations interact in standard notation.

Slurs and Ties

Now that you understand part writing in the **Scot** language, we can start discussing more elaborate features. Immediately following the pitch specification of each note, one may indicate a slur or a tie into the next note (assuming there is one), but not both simultaneously. The slur is typed as a single underscore (`_') and a tie as a double underscore (`__'). Despite the surface similarity, there is a substantial difference in the effect of these modifiers.

For purposes of **Scot**, tied notes are notes which, although comprised of several graphic symbols, represent only a single musical event. (Tied notes are necessary in standard music notation for several reasons, the most common being notes which span a measure line and notes with durations not specifiable with a single symbol, such as quarter note tied to a sixteenth.) Notes which are tied together are summed by duration and output by **Scot** as a single event. This means you cannot, for example, change the parameters of a note in the middle of a tie (see below). Two or more notes may be tied together, as in the following example, which plays an f# for eleven beats:

```
!ti "4/4"
1 f#__ / 1 f#__ / 2. f# 4r /
```

By contrast, slurred notes are treated as distinct notes at the **Csound** level, and may be of arbitrary pitch. The presence of a slur is reflected in parameter p4, but the slur has no other meaning beyond the interpretation of p4 by your instrument. Since instrument design is beyond the scope of this manual, it will suffice for now to explain that the **Scot** program gives sets p4 to one of four values depending on the existence of a slur before and after the note in question. This means **Scot** pays attention not only to the slur attached to a given note, but whether the preceding note specified a slur. The four possibilities are as follows, where the p4 values are taken to apply to the note `d':

```
4c  d       (no slur)            p4 = 0
4c  d_      (slur after only)    p4 = 1
4c_ d       (slur before only)   p4 = 2
4c_ d_      (before & after)     p4 = 3
```

Parameters

The information contained in the Scot score notation we have considered so far is manifested in the output score in parameters p1 through p5 in the following way:

p1: instrument number
p2: initialization time of instrument
p3: duration
p4: slur information
p5: pitch information in octave point pitch-class notation

Any additional parameters you may want to enter are listed in brackets as the last part of a note specification. These parameters start with p6 and are separated from each other with spaces. Any parameters not specified for a particular note have their value carried from the most recently specified value. You may choose to change some parameters and not others, in which case you can type a dot (`.') for parameters whose values don't change, and new values for those that do. Alternatively, the construction *N:*, where *N* is an integer, may be used to indicate that the following parameter specifications apply to successive parameters starting with parameter *N*. For example:

```
4e[15000 3 4 12:100 150] g_ d_[10000 . 5]    c
```

Here, for the first two quarter notes p6, p7, p8, p12, and p13 respectively assume the values 15000, 3, 4, 100, and 150. The values of p9 through p11 are either unchanged, or implicitly zero if they have never been specified in the current section. On the third quarter note, the value of p6 is changed to 10000, and the value of p8 is changed to 5. All others are unchanged.

Normally, parameter values are treated as globals—that is, a value specification will carry to successive notes if no new value is specified. However, if a parameter specification begins with an apostrophe, the value applies locally to the current note only, and will not carry to successive notes either horizontally or vertically (see *divisi* below).

Pfield Macros

Scot has a macro text substitution facility which operates only on the pfield specification text within brackets. It allows control values to be specified symbolically rather than numerically. Macro definitions appear inside brackets in the orchestra section. A single bracketed list of macro definitions preceding the first instrument declaration defines macros which apply to all instruments. An additional bracketed list of definitions may follow each instrument to specify macros that apply to that particular instrument.

```
orchestra {
        [ pp=2000 p=4000 mp=7000 mf=10000 f=20000 ff=30000
          modi = 11: w = 1 x = 2 y = 3 z = 4
          vib = "10:1 " novib = "10:0 1"
        ]
violin = 1      [ pizz = " 20:1" arco = " 20:0" ]
serpent = 3     [ ff = 25000 sfz = 'f sffz = 'ff]
}
score {
 $violin = 4c[mf modi z.y novib] d e a['f vib3] /
                8 b[pizz]c 4d[f] 2c[ff arco] /
 $serpent =, 4.c[mp modi y.x] 8b 2c /
                'g[f ], c[ff] /
}
```

As can be seen from this example, a macro definition consists of the macro name, which is a string of alphabetic characters, followed by an equal sign, followed by the macro value. As usual, spaces, tabs, and newlines may be used freely. The macro value may contain arbitrary characters, and may be quoted if spacing characters need to be included.

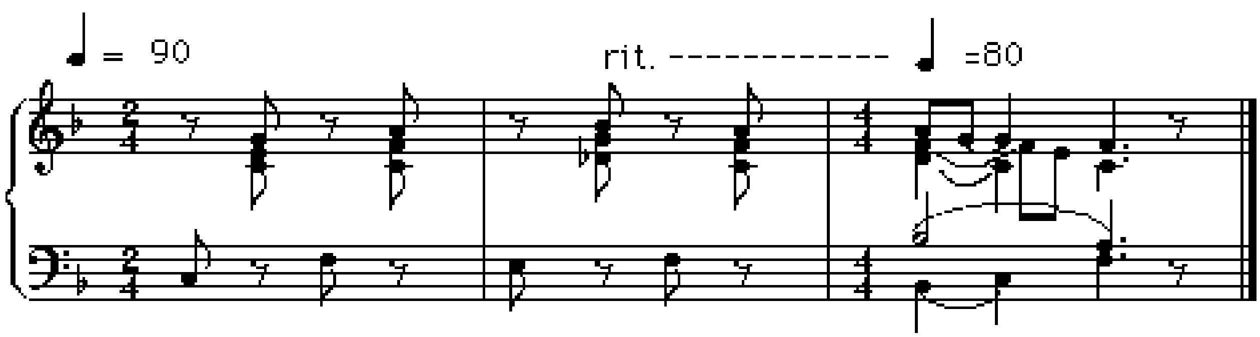
When a macro name is encountered in bracketed pfield lists in a score section, that name is replaced with the macro text with no additional punctuation supplied. The macro text may itself invoke other macros, although it is a serious error for a macro to contain itself, directly or indirectly. Since macro names are identified as strings of alphabetic characters, and no additional spaces are provided when a macro is expanded, macros may easily perform such concatenations as found in the first serpent note above, where the integer and fractional parts of a single pfield are constructed. Also, a macro may do no more than define a symbolic pfield, as in the definition of modi. The primary intention of macros is in fact not only to reduce the number of characters required, but also to enable symbolic definitions of parameter numbers and parameter values. For instance, a particular instrument's interpretation of the dynamic *ff* can be changed merely by changing a macro value, rather than changing all occurrences of that particular value in the score.

Divisi

Notes may be stacked on top of each other by using a back arrow (`<') between the notes of the divisi. Each time **Scot** encounters a back arrow, it understands that the following note is to start at the same time as the note to the left of the back arrow. Durations, accidentals and parameters carry from left to right through the divisi. Each time these are given new values, the notes to the right of the back arrows also take on the new values unless they are specified again.

When writing divisi you can stack compound events by enclosing them in parentheses. Also, divisi which occur at the end of the measure must have the proper durations or the scot program will mis-interpret the measure duration length.

Scot Example II



Scot encoding:

```
orchestra { right=1 left=2 }
functions { f1 0 256 10 1}
score {
$right !key "-b"
; since p5 is pitch, p7 is set to the pitch of next note
!ti "2/4"
!next p5 "p7"   ;since p5 is pitch, p7 refers to pitch of next note
!next p6 "p8"   ;If p6 is vol, say, then p8 refers to vol of next note
t90
8r c[3 np5]<e<='g r c<f<a / t90 r d-<g<b r =c[5]<f<a__ /
!ti "4/4"
t80
4d_<f__<(8a g__) 4c<(8fe)<4g 4.c<f<f 8r/

$left  !key "-b"
!next p5 "p7"
!next p6 "p8"
!ti "2/4"
8=,c[3 np5] r f r/ e r f r/
!ti "4/4"
2b_[5]<(4=,b_c) 4.a<f 8r/
}
```

Notice in this example that tempo statements occurred in instrument `right' only. Also, all notes had p6=3 until the third measure, at which point p6 took on the value 5 for all notes. The next parameter option used is described in Additional Features. The output score is given at the end.

Additional Features

Several options can be included in any of the individual instrument parts within a section. A sample statement follows the description of each option. The keyword which follows the `!' in these statements may be abbreviated to the first two characters.

Key Signatures

Any desired key signature is specified by listing the accidentals as they occur in a key signature statement. Thereafter, all notes of that instrument part are sharpened or flattened accordingly. For example, for the key of D, type

```
!key "#fc"
```

Accidental Following

Accidental following may be turned on or off as needed. When turned off, accidentals no longer carry throughout the measure as in traditional notation. This convention is sometimes used in contemporary scores.

```
!accidentals "off"
```

Octave Following

Turning off octave following indicates that pitches stay in the same absolute octave register until explicitly moved. An absolute octave starts at pitch c and ends at the b above it. The octave middle-c-to-b is indicated with an equals sign (`=') and octave displacement is indicated with the appropriate number of commas or apostrophes. These displacements are cummulative. For example,

```
!octaves "off"
4='c g b 'c
```

starts at the c above middle c and ends at two c's above middle c.

Vertical Following

Turning off vertical following means that durations, register, and parameters only carry horizontally from note to note and not vertically as described in the section on divisi.

```
!vertical "off"
```

Transposition

Any instrument part can be transposed to another key by indicating the intervalic difference between the notated key and the desired key. This difference is always taken with reference to middle c - to transpose a whole step upward, for example, type

```
!transpose "d"
```

This indicates that the part is transposed by the interval difference between middle c and d.

Next-value and Previous-value Parameteres

In order to play a note, it is sometimes necessary for an instrument to know what value one or more parameters will have for the next note. For instance, an instrument might be designed which glisses during the last portion of its performance (perhaps only when a slur is indicated) from its written pitch to the pitch of the next note. This can only be done, of course, if the instrument can know what the pitch of the next note will be.

The necessary information can be provided using next-value parameters. A next value parameter might be declared by

```
!next p5 "p6"
```

which is interpreted to mean that for the current instrument, p6 will contain the next note's p5 value. This holds true globally for all occurences of this instrument until further modifications. If for any reason you wish to override this value, p6 may be filled in explicitly. This is sometimes useful for the last note of a section, for which p6 will otherwise assume the p5 value for the current note. The *next-value* feature is illustrated in the Scot example II.

The necessary information may also be provided using *standard numeric score* next-value parameters. A parameter argument containing the symbol npx (where x is an integer) will substitute parameter number x of the following note for that instrument. Similarly, the value of a parameter occurring during the previous note may be referenced with the symbol ppx (where x is an integer). Details of the next- and previous-value parameter feature may be found in the **Numeric Scores** section.

Ramping

Pfields containing the symbol < will be replaced by values derived from linear interpolation of a time-based ramp. Ramp endpoints are defined by the first real number found in the same pfield of a preceding and following note played by the same instrument. Details of the ramping feature are likewise found in the **Numeric Scores** section.

f0 Statements

In each score section, **Scot** automatically produces an f0 statement with a p2 equal to the ending time of the last note or rest in the section. Thus, `dead time' at the end of a section for reverberation decay or whatever purpose may be specified musically by rests in one or more parts. See the eighth rest at the end of Scot example II and its output score shown below.

Output Scores

Output file *score* from Scot Example I.

```
f1 0 512 10 1 .5 .25 .126
f2 0 256  7 1 120 1 8 0 128 1
i1.01 0 1 0 8.00
i1.01 1 0.5 0 8.02
i1.01 1.5 0.5 0 8.03
i1.01 2 0.5 0 8.05
i1.01 3 1 0 9.00
i1.01 4 0.75 0 8.10
i1.01 4.75 0.25 0 8.09
i1.01 5 0.25 0 8.08
i1.01 5.25 0.25 0 8.07
i1.01 5.5 0.25 0 8.06
i1.01 5.75 0.25 0 8.05
i1.01 6 1 0 8.03
i1.01 7 1 0 8.00
i2.01 0 2 0 6.00
i2.01 2 2 0 6.08
i2.01 4 2 0 6.07
i2.01 6 2 0 7.00
t0 60
f0 8
s
i1.01 0 0.6667 0 9.00
i1.01 1.3333 0.6667 0 9.00
i1.01 2 1.75 0 9.02
i1.01 3.75 0.25 0 9.03
i1.01 4 0.6667 0 9.05
i1.01 5.3333 0.6667 0 9.05
i1.01 6 1.75 0 9.07
i1.01 7.75 0.25 0 9.09
i2.01 0 1 0 6.00
i2.01 1 1 0 6.07
i2.01 2 1 0 6.00
i2.01 3 1 0 6.07
i2.01 4 2 0 7.08
i2.01 6 2 0 7.07
t0 60
f0 8
s
```

Output file *score* from Scot Example II.

```
f1 0 256 10 1
c r1 n 7 5
c r1 n 8 6
i1.01 0.5000 0.5000 0 8.00 3 8.00 3
i1.02 0.5000 0.5000 0 8.04 3 8.05 3
i1.03 0.5000 0.5000 0 8.07 3 8.09 3
i1.01 1.5000 0.5000 0 8.00 3 8.01 3
i1.02 1.5000 0.5000 0 8.05 3 8.07 3
i1.03 1.5000 0.5000 0 8.09 3 8.10 3
i1.01 2.5000 0.5000 0 8.01 3 8.00 5
i1.02 2.5000 0.5000 0 8.07 3 8.05 5
i1.03 2.5000 0.5000 0 8.10 3 8.09 5
i1.01 3.5000 0.5000 0 8.00 5 8.02 5
i1.02 3.5000 0.5000 0 8.05 5 8.05 5
i1.01 4.0000 1.0000 1 8.02 5 8.00 5
i1.03 3.5000 1.0000 0 8.09 5 8.07 5
i1.01 5.0000 1.0000 2 8.00 5 8.00 5
i1.02 4.0000 1.5000 0 8.05 5 8.04 5
i1.02 5.5000 0.5000 0 8.04 5 8.05 5
i1.03 4.5000 1.5000 0 8.07 5 8.05 5
i1.01 6.0000 1.5000 0 8.00 5 8.00 5
i1.02 6.0000 1.5000 0 8.05 5 8.05 5
i1.03 6.0000 1.5000 0 8.05 5 8.05 5
c r2 n 7 5
c r2 n 8 6
i2.01 0.0000 0.5000 0 7.00 3 7.05 3
i2.01 1.0000 0.5000 0 7.05 3 7.04 3
i2.01 2.0000 0.5000 0 7.04 3 7.05 3
i2.01 3.0000 0.5000 0 7.05 3 7.10 5
i2.01 4.0000 2.0000 1 7.10 5 7.09 5
i2.02 4.0000 1.0000 1 6.10 5 7.00 5
i2.02 5.0000 1.0000 2 7.00 5 7.05 5
i2.01 6.0000 1.5000 2 7.09 5 7.09 5
i2.02 6.0000 1.5000 0 7.05 5 7.05 5
t0 60 0.0000 90.0000 2.0000 90.0000 4.0000 80.0000 4.0000 90.0000
f0 8.0000
s
e
```

6. The Unix CSOUND Command

csound is a command for passing an orchestra file and score file to **Csound** to generate a soundfile. The score file can be in one of many different formats, according to user preference. Translation, sorting, and formatting into orchestra-readable numeric text is handled by various preprocessors; all or part of the score is then sent on to the orchestra. Orchestra performance is influenced by command flags, which set the level of displays and console reports, specify 1/0 filenames and sample formats, and declare the nature of realtime sensing and control.

The format of a command is:

csound [-flags] orchname scorename

where the arguments are of 2 types: *flag* arguments (beginning with a "-"), and *name* arguments (such as filenames). Certain flag arguments take a following name or numeric argument. The available flags are:

-I, -n	sound output inhibitors
-iName, -oName	sound I/O filenames
-bNumb, -BNumb, -h	audio buffers & header control
-A, -c, -a, -u, -s, -1, -f	audio output formats
-v, -mNumb, -d, -g	message & display levels
-S, -xName, -tNumb	score formats & tempo control
-LName	line-oriented realtime event stream
-MName, -FName, -PNumb	MIDI event streams
-N, -T	notify or terminate when done

Flags may appear anywhere in the command line, either separately or bundled together. A flag taking a Name or Number will find it in that argument, or in the immediately subsequent one. The following are thus equivalent commands:

csound -nm3 orchname -Sxxfilename scorename
csound -n -m 3 orchname -x xfilename -S scorename

All flags and names are optional. The default values are:

csound -s -otest -b1024 -B1024 -m7 -P128 orchname scorename

where orchname is a file containing Csound orchestra code,

and scorename is a file of score data in standard numeric score format, optionally presorted and time-warped. If scorename is *omitted*, there are two default options: 1) if realtime input is expected (-L, -M or -F), a dummy scorefile is substituted consisting of the single statement 'f 0 3600' (i.e. listen for RT input for one hour); 2) else **csound** uses the previously processed *score.srt* in the *current directory*.

Csound reports on the various stages of score and orchestra processing as it goes, doing various syntax and error checks along the way. Once the actual performance has begun, any error messages will derive from either the instrument loader or the unit generators themselves. A **csound** command may include any rational combination of the following flag arguments, with default values as described:

csound -I
I-time only. Allocate and initialize all instruments as per the score, but skip all P-time processing (no k-signals or a-signals, and thus no amplitudes and no sound). Provides a fast validity check of the score pfields and orchestra **i**-variables.

csound -n
Nosound. Do all processing, but bypass writing of sound to disk. This flag does not change the execution in any other way.

csound -i isfname
Input soundfile name. If not a full pathname, the file will be sought first in the current directory, then in that given by the environment variable SSDIR (if defined), then by SFDIR. The name *stdin* will cause audio to be read from standard input. If RTAUDIO is enabled, the name *devaudio* will request sound from the host audio input device.

csound -o osfname
Output soundfile name. If not a full pathname, the soundfile will be placed in the directory given by the environment variable SFDIR (if defined), else in the current directory. The name *stdout* will cause audio to be written to standard output. If no name is given, the default name will be test. If RTAUDIO is enabled, the name *devaudio* will send to the host audio output device.

csound -b Numb
Number of audio sample-frames per soundio *software* buffer. Large is efficient, but small will reduce audio I/O delay. The default is 1024. In **realtime performance, Csound** waits on audio I/O on *Numb* boundaries. It also processes audio (and polls for other input like MIDI) on orchestra *ksmps* boundaries. The two can be made synchronous. For convenience, if Numb = -N (is negative) the effective value is *ksmps * N* (audio synchronous with k-period boundaries). With N small (e.g. 1) polling is then frequent and also locked to fixed DAC sample boundaries.

csound -B Numb
Number of audio sample-frames held in the DAC *hardware* buffer. This is a threshold on which *software* audio I/O (above) will wait before returning. A small number reduces audio I/O delay; but the value is often hardware limited, and small values will risk data lates. The default is 1024.

csound -h
No header on output soundfile. Don't write a file header, just binary samples.

csound {-c, -a, -u, -s, -l, -f}
Audio sample format of the output soundfile. One of:

c	8-bit signed character
a	8-bit a-law
u	8-bit u-law
s	short integer
l	long integer
f	single-precision float (not playable, but can be read by -i, **soundin** and GEN01)

csound -A
Write an AIFF output soundfile. Restricts the above formats to **c, s,** or **l.**

csound -v
Verbose translate and run. Prints details of orch translation and performance, enabling errors to be more clearly located.

csound -m Numb
Message level for standard (terminal) output. Takes the *sum* of 3 print control flags, turned on by the following values: 1 = note amplitude messages, 2 = samples out of range message, 4 = warning messages. The default value is *m7* (all messages on).

csound -d
Suppress all displays.

csound -g
Recast graphic displays into *ascii* characters, suitable for any terminal.

csound -S
Interpret *scorename* as a Scot file and create a standard score file (named "score") from it, then sort and perform that.

csound -x xfile
Extract a portion of the sorted score *score.srt*, according to *xfile* (see extract below).

csound -t Numb
Use the uninterpreted beats of *score.srt* for this performance, and set the initial tempo at *Numb* beats per minute. When this flag is set, the tempo of score performance is also controllable from within the orchestra (see the **tempo** unit).

csound -L devname
Read Line-oriented realtime score events from device *devname*. The name *stdin* will permit score events to be typed at your terminal, or piped from another process. Each line-event is terminated by a carriage-return. Events are coded just like those in a *standard numeric score*, except that an event with p2=0 will be performed immediately, and an event with p2=T will be performed T seconds after arrival. Events can arrive at any time, and in any order. The score *carry feature* is legal here, as are held notes (p3 negative) and string arguments, but ramps and pp or np references are not.

csound -M devname
Read MIDI events from device *devname*.

csound -F mfname
Read MIDI events from midifile *mfname*.

csound -P Numb
Set MIDI sustain pedal threshold (0 - 128). The official switch value of 64 is normally too low, and is more realistic above 100. The default value of 128 will block all pedal info.

csound -N
Notify (ring the bell) when score or miditrack is done.

csound -T
Terminate the performance when miditrack is done.

The EXTRACT feature

This feature will **extract** a segment of a sorted numeric score file according to instructions taken from a control file. The control file contains an instrument list and two time points, *from* and *to*, in the form:

instruments 1 2 from 1:27.5 to 2:2

The component labels may be abbreviated as *i*, *f* and *t*. The time points denote the beginning and end of the extract in terms of:

[section no.] : [beat no.].

each of the three parts is also optional. The default values for missing *i*, *f* or *t* are:

all instruments, beginning of score, end of score.

extract reads an orchestra-readable score file and produces an orchestra-readable result. Comments, tabs and extra spaces are flushed, **w** and **a** statements are added and an **f0** reflecting the extract length is appended to the output. Following an **extract** process, the abbreviated score will contain all function table statements, together with just those note statements that occur in the *from-to* interval specified. Notes lying completely in the interval will be unmodified; notes that lie only partly within will have their p3 durations truncated as necessary.

Independent Preprocessing

Although the result of all score preprocessing is retained in the file *score.srt* after orchestra performance (it exists as soon as score preprocessing has completed), the user may sometimes want to run these phases independently. The command

scot filename

will process the Scot formatted *filename*, and leave a *standard numeric score* result in a file named *score* for perusal or later processing.

The command

scscort < infile > outfile

will put a numeric score *infile* through Carry, Tempo, and Sort preprocessing, leaving the result in *outfile*.

Likewise extract, also normally invoked as part of the **csound** command, can be invoked as a standalone program:

extract xfile < score.sort > score.extract

This command expects an already sorted score. An unsorted score should first be sent through *scsort* then piped to the extract program:

scsort < scorefile | extract xfile > score.extract

Appendix 1: CSCORE

Cscore is a standalone program for generating and manipulating numeric score files. It comprises a number of *function* subprograms, called into operation by a user-written *main* program. The *function* programs augment the **C** language library functions; they can optionally read *standard numeric* score files, can massage and expand the data in various ways, then write the data out as a new score file to be read by a **Csound** orchestra.

The user-written *main* program is also in **C**. It is not essential to know the **C** language well to write a main program, since the function calls have a simple syntax, and are powerful enough to do most of the complicated work. Additional power can come from **C** later as the need arises.

Events, Lists, and Operations

An *event* in **Cscore** is equivalent to one statement of a *standard numeric score*. It is either created or read in from an existing score file. An event is comprised of an *opcode* and an array of *pfield* values stored somewhere in memory. Storage is organized by the following structure:

```
struct event {
        char op;                /* opcode */
        char tnum;
        short pcnt;
        float p[PMAX+1];        /* pfields */
};
```

Any function subprogram that creates, reads, or copies an event function will return a *pointer* to the storage structure holding the event data. The event pointer can be used to access any component of the structure, in the form of *e->op* or *e->p[n]*. Each newly stored event will give rise to a new pointer, and a sequence of new events will generate a sequence of distinct pointers that must themselves be stored. Groups of event pointers are stored in a *list*, which has its own structure:

```
struct evlist {
        int nslots;             /* size of this list      */
        struct event *e[1];     /* list of event pointers */
};
```

Any function that creates or modifies a *list* will return a *pointer* to the new list. The list pointer can be used to access any of its component event pointers, in the form of *a->e[n]*. Event pointers and list pointers are thus primary tools for manipulating the data of a score file.

Pointers and *lists of pointers* can be copied and reordered without modifying the data values they refer to. This means that notes and phrases can be copied and manipulated from a high level of control. Alternatively, the data within an *event* or *group of events* can be modified without changing the event or list pointers. **Cscore** provides a library of programming *methods* or *function subprograms* by which scores can be created and manipulated in this way.

In the following summary of **Cscore** function calls, some simple naming conventions are used:

> the symbols *e, f* are pointers to events (notes);
> the symbols *a, b* are pointers to lists (arrays) of such events;
> the letters *ev* at the end of a function name signify operation on an *event*;
> the letter *l* at the start of a function name signifies operation on a *list*.

calling syntax	description
e = createv(n);	create a blank event with n pfields
e = defev("...");	defines an event as per the character string ...
e = copyev(f);	make a new copy of event f
e = getev();	read the next event in the score input file
putev(e);	write event e to the score output file
putstr("...");	write the character string ... to score output
a = lcreat(n);	create an empty event list with n slots
a = lappev(a,e);	append event e to list a
n = lcount(a);	count the events now in list a
a = lcopy(b);	copy the list b (but not the events)
a = lcopyev(b);	copy the events of b, making a new list
a = lget();	read events from score input (to next s or e)
lput(a);	write the events of list a to score output
a = lsepf(b);	separate the f statements from list b into list a
a = lcat(a,b);	concatenate (append) the list b onto the list a
lsort(a);	sort the list a into chronological order by p[2]
a = lxins(b,"...");	extract notes of instruments ... (no new events)
a = lxtimev(b,from,to);	extract notes of time-span, creating new events
relev(e);	release the space of event e
lrel(a);	release the space of list a (but not events)
lrelev(a);	release the events of list a, and the list space

Writing a Main program.

The general format for a main program is:

```
#include  <csound/cscore.h>
main()
{
        /*  VARIABLE DECLARATIONS     */

        /*  PROGRAM BODY              */
}
```

The *include* statement will define the event and list structures for the program. The following **C** program will read from a standard numeric score, up to (but not including) the first **s** or **e** statement, then write that data (unaltered) as output.

```
#include  <csound/cscore.h>
main()
{
        struct evlist *a;        /* a is allowed to point to an event list */

        a = lget();              /* read events in, return the list pointer */
        lput(a);                 /* write these events out (unchanged) */
        putstr("e");             /* write the string e to output */
}
```

After execution of *lget()*, the variable *a* points to a list of event addresses, each of which points to a stored event. We have used that same pointer to enable another list function (*lput*) to access and write out all of the events that were read. If we now define another symbol *e* to be an *event* pointer, then the statement

```
e = a->e[4];
```

will set it to the *contents* of the 4th slot in the evlist structure. The contents is a pointer to an event, which is itself comprised of an array of parameter field values. Thus the term e->p[5] will mean the value of parameter field 5 of the 4th event in the evlist denoted by *a*. The program below will multiply the value of that pfield by 2 before writing it out.

```
#include  <csound/cscore.h>
main()
{
        struct event  *e;        /* a pointer to an event                  */
        struct evlist *a;

        a = lget();              /* read a score as a list of events       */
        e = a->e[4];             /* point to event 4 in event list \fIa\fR */
        e->p[5] *= 2;            /* find pfield 5, multiply its value by 2 */
        lput(a);                 /* write out the list of events           */
        putstr("e");             /* add a "score end" statement            */
}
```

Now consider the following score, in which p[5] contains frequency in cps.

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
e
```

If this score were given to the preceding main program, the resulting output would look like this:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
i 1 1 3 0 440 10000
i 1 4 3 0 512 10000        ; p[5] has become 512 instead of 256.
i 1 7 3 0 880 10000
e
```

Note that the 4th event is in fact the second note of the score. So far we have not distinguished between notes and function table setup in a numeric score. Both can be classed as events. Also note that our 4th event has been stored in e[4] of the structure. For compatibility with **Csound** pfield notation, we will ignore p[0] and e[0] of the event and list structures, storing p1 in p[1], event 1 in e[1], etc. The **Cscore** functions all adopt this convention.

As an extension to the above, we could decide to use *a* and *e* to examine each of the events in the list. Note that *e* has not preserved the numeral 4, but the contents of that slot. To inspect p5 of the previous listed event we need only redefine *e* with the assignment

```
e = a->e[3];
```

More generally, if we declare a new variable *f* to be a *pointer to a pointer to an event*, the statement

```
f = &a->e[4];
```

will set *f* to the address of the fourth event in the event list *a*, and **f* will signify the *contents* of the slot, namely the event pointer itself. The expression

```
(*f)->p[5],
```

like *e->p[5],* signifies the fifth pfield of the selected event. However, we can advance to the next slot in the evlist by advancing the pointer *f*. In **C** this is denoted by *f++*.

In the following program we will use the same input score. This time we will separate the ftable statements from the note statements. We will next write the three note-events stored in the list *a*, then create a second score section consisting of the original pitch set and a transposed version of itself. This will bring about an octave doubling.

By pointing the variable *f* to the first note-event and incrementing *f* inside a **while** block which iterates n times (the number of events in the list), one statement can be made to act upon the same pfield of each successive event.

```
#include  <csound/cscore.h>
main()
{
        struct event *e,**f;            /* declarations. see pp.8-9 in the */
        struct evlist *a,*b;            /* C language programming manual */
        int n;

        a = lget();                     /* read score into event list "a" */
        b = lsepf(a);                   /* separate f statements */
        lput(b);                        /* write f statements out to score */
        lrelev(b);                      /* and release the spaces used */
        e = defev("t 0 120");           /* define event for tempo statement */
        putev(e);                       /* write tempo statement to score */
        lput(a);                        /* write the notes */
        putstr("s");                    /* section end */
        putev(e);                       /* write tempo statement again */
        b = lcopyev(a);                 /* make a copy of the notes in "a" */
        n = lcount(b);                  /* and count the number copied */
        f = &a->e[1];
        while (n--)                     /* iterate the following line n times: */
           (*f++)->p[5] *= .5;          /*   transpose pitch down one octave */
        a = lcat(b,a);                  /* now add these notes to original pitches */
        lput(a);
        putstr("e");
}
```

The output of this program is:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
t 0 120
i 1 1 3 0 440 10000
```

```
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
s
t 0 120
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
i 1 1 3 0 220 10000
i 1 4 3 0 128 10000
i 1 7 3 0 440 10000
e
```

Next we extend the above program by using the **while** statement to look at p[5] and p[6]. In the original score p[6] denotes amplitude. To create a diminuendo in the added lower octave, which is independent from the original set of notes, a variable called *dim* will be used.

```
#include <csound/cscore.h>
main()
{
        struct event *e,**f;
        struct evlist *a,*b;
        int n, dim;                     /* declare new variable as integer */

        a = lget();
        b = lsepf(a);
        lput(b);
        lrelev(b);
        e = defev("t 0 120");
        putev(e);
        lput(a);
        putstr("s");
        putev(e);                       /* write out another tempo statement */
        b = lcopyev(a);
        n = lcount(b);
        dim = 0;                        /* initialize dim to 0 */
        f = &a->e[1];
        while (n--){
            (*f)->p[6] -= dim;          /* subtract current value of dim */
            (*f++)->p[5] *= .5;         /* transpose, move f to next event */
            dim += 2000;                /* increase dim for each note */
        }
        a = lcat(b,a);
        lput(a);
        putstr("e");
}
```

The increment of *f* in the above programs has depended on certain precedence rules of **C**. Although this keeps the code tight, the practice can be dangerous for beginners. Incrementing may alternately be written as a separate statement to make it more clear.

```
        while (n--){
            (*f)->p[6] -= dim;
            (*f)->p[5] *= .5;
            dim += 2000;
```

```
        f++;
    }
```

Using the same input score again, the output from this program is:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
t 0 120
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000
s
t 0 120
i 1 1 3 0 440 10000            ; Three original notes at
i 1 4 3 0 256 10000            ; beats 1,4 and 7 with no dim.
i 1 7 3 0 880 10000
i 1 1 3 0 220 10000            ; three notes transposed down one octave
i 1 4 3 0 128 8000             ; also at beats 1,4 and 7 with dim.
i 1 7 3 0 440 6000
e
```

In the following program the same three-note sequence will be repeated at various time intervals. The starting time of each group is determined by the values of the array *cue*. This time the *dim* will occur for each group of notes rather than each note. Note the position of the statement which increments the variable *dim* outside the inner **while** block.

```
#include  <csound/cscore.h>

int cue[3]={0,10,17};                    /* declare array of 3 integers */

main()
{
    struct event *e, **f;
    struct evlist *a, *b;
    int n, dim, cuecount, holdn;         /* declare new variables */

    a = lget();
    b = lsepf(a);
    lput(b);
    lrelev(b);
    e = defev("t 0 120");
    putev(e);
    n = lcount(a);
    holdn = n;                           /* hold the value of "n" to reset below */
    cuecount = 0;                        /* initilize cuecount to "0" */
    dim = 0;
    while (cuecount <= 2) {              /* count 3 iterations of inner "while" */
        f = &a->e[1];                    /* reset pointer to first event of list "a" */
        n = holdn;                       /* reset value of "n" to original note count */
        while (n--) {
            (*f)->p[6] -= dim;
            (*f)->p[2] += cue[cuecount];         /* add values of cue */
            f++;
        }
```

```
                printf("; diagnostic:  cue = %d\n", cue[cuecount]);
                cuecount++;
                dim += 2000;
                lput(a);
            }
            putstr("e");
    }
```

Here the inner **while** block looks at the events of list *a* (the notes) and the outer **while** block looks at each repetition of the events of list *a* (the pitch group repetitions). This program also demonstrates a useful trouble-shooting device with the **printf** function. The semi-colon is first in the character string to produce a comment statement in the resulting score file. In this case the value of *cue* is being printed in the output to insure that the program is taking the proper array member at the proper time. When output data is wrong or error messages are encountered, the **printf** function can help to pinpoint the problem.

Using the identical input file, the **C** program above will generate:

```
f 1 0 257 10 1
f 2 0 257 7 0 300 1 212 .8
t 0 120

; diagnostic:  cue = 0
i 1 1 3 0 440 10000
i 1 4 3 0 256 10000
i 1 7 3 0 880 10000

; diagnostic:  cue = 10
i 1 11 3 0 440 8000
i 1 14 3 0 256 8000
i 1 17 3 0 880 8000

; diagnostic:  cue = 17
i 1 28 3 0 440 4000
i 1 31 3 0 256 4000
i 1 34 3 0 880 4000
e
```

Further development of these scores will lead the composer to techniques of score manipulation which are similar to serial techniques of composition. Pitch sets may be altered with regard to any of the parameter fields. The programing allows transpositions, time warping, pitch retrograding and dynamic controls, to name a few.

Compiling a Cscore program

A **Cscore** program *example.c* is compiled and linked with its library modules by the command:

```
$ cc  -o myprog  example.c  -lcscore
```

The resulting executable file *myprog* is run by typing:

```
$ myprog                          (no input, output printed on the screen)
$ myprog < scorin                 (input score named \fIscorin\fR, output on screen)
$ myprog < scorin  > scorout      (input as above, output into a file)
```

Appendix 2: An Instrument Design Tutorial

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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 various Csound directories, and a study of these will reveal Csound's real power.

The Csound **orchestra** file has two main parts:

1. the “header” section - defining the sample rate, control rate, and number of output channels.
2. 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 16kHz, a control rate of 1kHz, and then calculated the number of samples in each control period using the formula: ksmps = sr / kr.

```
sr      =       16000
kr      =       1000
ksmps   =       16
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** 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 waveshape (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.

Tutorial Instruments

Toot 1: Play One Note

For this and all instrument examples below, there exist **orchestra** and **score** files in the Csound subdirectory **tutorfiles** that the user can run to soundtest each feature introduced. The instrument code shown below is actually preceded by an *orchestra header section* similar to that shown above. If you are running on a RISC computer, each example will likely run in realtime. During playback (realtime or otherwise) the audio rate may automatically be modified to suit the local d-a converters.

The first orchestra file, called *toot1.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
```

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

```
f1   0    4096   10   1     ; use "gen1" to compute a sine wave
i1   0    4                 ; run "instr 1" from time 0 for 4 seconds
e                           ; indicate the "end" of the score
```

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

      f1    0       4096    10      1                   ; sine wave
; instrument start   duration  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
```

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

  f1  0  4096  10  1                        ; sine wave
; instr  start  duration  amp(p4)  freq(p5)  attack(p6)  release(p7)
  i3     0      1         10000    440       .05         .7
  i3     1.5    1         10000    440       .9          .1
  i3     3      1          5000    880       .02         .99
  i3     4.5    1          5000    880       .7          .01
  i3     6      2         20000    220       .5          .5
e
```

Toot 4: Chorusing

Next we'll animate the basic sound by mixing it with two slightly detuned 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                                      ; toot4.orc
  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
```

```
  f1 0 4096 10 1                                               ; sine wave
; instr   start  duration  amp(p4)   freq(p5)  attack(p6)  release(p7)
  i4      0      1         75        8.04      .1          .7
  i4      1      1         70        8.02      .07         .6
  i4      2      1         75        8.00      .05         .5
  i4      3      1         70        8.02      .05         .4
  i4      4      1         85        8.04      .1          .5
  i4      5      1         80        8.04      .05         .5
  i4      6      2         90        8.04      .03         1
e
```

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                                       ; toot5.orc
          irel = .01                                  ; set vibrato release time
          idel1 = p3 - (p10 * p3)                     ; 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
```

```
  f 1  0  4096  10  1
; ins   strt   dur   amp   frq     atk   rel   vibrt   vibdpth  vibdel
  i5    0      3     86    10.00   .1    .7    7       6        .4
  i5    4      3     86    10.02   1     .2    6       6        .4
  i5    8      4     86    10.04   2     1     5       6        .4
e
```

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.

F-statements, or lines with an opcode of **f**, invoke function-drawing subroutines called **GENS**. In Csound there are currently seventeen 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  .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
```

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 -> p? - 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                                          ; toot6.orc
  ifunc = p11                                    ; select basic waveform
  irel = .01                                     ; set vibrato release
  idel1 = p3 - (p10 * p3)                        ; calculate initial delay
  isus   = p3 - (idel1- irel)                    ; calculate remaining dur
  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
```

```
  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   vibrt  vibdpth  vibdel  waveform(f)
  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      4
e
```

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                                          ; toot7.orc
  ifunc1 = p11                                   ; initial waveform
  ifunc2 = p12                                   ; crossfade waveform
  ifad1 = p3 - (p13 * p3)                        ; calculate initial fade
  ifad2 = p3 - ifad1                             ; calculate remaining dur
  irel = .01                                     ; set vibrato release
  idel1 = p3 - (p10 * p3)                        ; calculate initial delay
  isus   = p3 - (idel1- irel)                    ; calculate remaining dur
  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
```

```
 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 vibrt vbdpt vibdel startwave  endwave  crosstime
 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        .1
```

Toot 8: Soundin

Now instead of continuing to enhance the same instrument, let us design a totally different one. 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 **allpass** 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                                                  ; toot8.orc
  idur          =            p3
  iamp          =            p4
  iskiptime     =            p5
  iattack       =            p6
  irelease      =            p7
  irvbtime      =            p8
  irvbgain      =            p9
  kamp          linen        iamp, iattack, idur, irelease
  asig          soundin      "soundin.aiff", iskiptime
  arampsig       =           kamp * asig
  aeffect       reverb       asig, irvbtime
  arvbreturn    =            aeffect * irvbgain
                out          arampsig + arvbreturn
endin
```

```
; ins  strt    dur    amp   skip    atk    rel    rvbtime  rvbgain
  i8   0       1      .3    0       .03    .1     1.5      .2
  i8   2       1      .3    0       .1     .1     1.3      .2
  i8   3.5     2.25   .3    0       .5     .1     2.1      .2
  i8   4.5     2.25   .3    0       .01    .1     1.1      .2
  i8   5       2.25   .3    .1      .01    .1     1.1      .1
e
```

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 instr99 below receives input from instr9 via the global a-rate variable *garvbsig*. Since instr9 *adds into* this global, several copies of instr9 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 instr99 once its data has been placed into the reverb.

```
sr          =       18900          ; toot9.orc
kr          =       945
ksmps       =       20
nchnls      =       2              ; stereo
garvbsig    init    0              ; make zero at orch init time

instr 9
  idur        =       p3
  iamp        =       p4
  iskiptime   =       p5
  iattack     =       p6
  irelease    =       p7
  ibalance    =       p8           ; panning: 1=left, .5=center, 0=right
  irvbgain    =       p9
  kamp        linen        iamp, iattack, idur, irelease
  asig        soundin      "soundin.aiff", iskiptime
  arampsig    =            kamp * asig
              outs         arampsig * ibalance,  arampsig * (1 - ibalance)
  garvbsig    =            garvbsig + arampsig * irvbgain
endin
```

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

In the score we turn the global reverb on at time 0 and keep it on until *irvbtime* after the last note.

```
; ins   strt    dur     rvbtime                         ; toot9.sco
 i99    0       9.85    2.6

; ins   strt    dur     amp  skip   atk     rel     balance(0-1)   rvbsend
  i9    0       1       .5    0     .02     .1      1              .2
  i9    2       2       .5    0     .03     .1      0              .3
  i9    3.5     2.25    .5    0     .9      .1      .5             .1
  i9    4.5     2.25    .5    0     1.2     .1      0              .2
  i9    5       2.25    .5    0     .2      .1      1              .3
e
```

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 above.

```
garvbsig        init    0

instr 10                                ; toot10.orc
  iattack       =       .01
  irelease      =       .2
  iwhite        =       10000
  idur          =       p3
  iamp          =       p4
  isweepstart   =       p5
  isweepend     =       p6
  ibandwidth    =       p7
  ibalance      =       p8              ; pan: 1 = left, .5 = center, 0 = right
  irvbgain      =       p9
  kamp          linen   iamp, iattack, idur, irelease
  ksweep        line    isweepstart, idur, isweepend
  asig          rand    iwhite
  afilt         reson   asig, ksweep, ibandwidth
  arampsig      =       kamp * afilt
                outs    arampsig * ibalance, arampsig * (1 - ibalance)
  garvbsig      =       garvbsig  +  arampsig * irvbgain
endin

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

```
; ins   strt   dur   rvbtime                              ; toot10.sco
  i100  0      15    1.1
  i100  15     10    5

; ins   strt   dur   amp    stswp  ndswp  bndwth  balance(0-1)  rvbsend
  i10   0      2     .05    5000   500    20      .5            .1
  i10   3      1     .05    1500   5000   30      .5            .1
  i10   5      2     .05    850    1100   40      .5            .1
  i10   8      2     .05    1100   8000   50      .5            .1
  i10   8      .5    .05    5000   1000   30      .5            .2
  i10   9      .5    .05    1000   8000   40      .5            .1
  i10   11     .5    .05    500    2100   50      .4            .2
  i10   12     .5    .05    2100   1220   75      .6            .1
  i10   13     .5    .05    1700   3500   100     .5            .2
  i10   15     5     .01    8000   800    60      .5            .15
e
```

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.

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:

```
;  ins  start  dur  amp  freq        |        ;  ins  start  dur  amp  freq
   i11  0      1    90   200         |           i11  0      1    90   200
   i11  +      2                     |           i11  1      2    90   200
   i11                               |           i11  3      2    90   200
```

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
 i11    0       1      90     200
 i .    +       .      <      <
 i .    .       .      <      400
 i .    .       .      <      <
 i .    .       4      70     200
```

```
; ins   start   dur    amp    freq
 i11    0       1      90     200
 i11    1       1      85     300
 i11    2       1      80     400
 i11    3       1      75     300
 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 (see the Csound manual). 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 soundfiles:

```
; ins   start   dur    amp    freq
 i11    0       .5     90     200
 i .    +       .      <      <
 i .    .       .      <      400
 i .    .       .      <      <
 i .    .       2      70     200
```

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

```
;       ins     start   dur     amp     freq                    ; toot11g.sco
        i11     0       2       90      100
        f 0     4                                               ; The f0 Statement
        s                                                       ; The Section Statement
        i11     0       1       90      800
        i .     +       .       .       400
        i .     .       .       .       100
        f 0     5
        s
        i11     0       4       90      50
        e
```

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   vibrt  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
```

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                      ; lfo and random index rate
              itab      =    2
              itablesize =   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  *  itablesize, 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 osc
                       out      asig
            endin

 f1 0 2048 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  start   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
s

; method 2 - lfo index of table values
; ins  start  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
s

; method 3 - random index of table values
; ins  start  dur   amp   index  method  rndrate  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      .1
```

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 most powerful unit generators—**pluck**, **fof**, and **foscil**, make this complex task a fairly simple one. The Reference Manual describes these as follows:

```
a1   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.

```
a2   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.

```
a1   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                                ; toot13.orc
          iamp          = ampdb(p4) / 2         ; amplitude, scaled for two sources
          ipluckamp     = p6                    ; % of total amp, 1=dB amp as in p4
          ipluckdur     = p7*p3                 ; % of total dur, 1=entire dur of note
          ipluckoff     = p3 - ipluckdur
          ifmamp        =  p8                   ; % of total amp, 1=dB amp as in p4
          ifmrise       = p9*p3                 ; % of total dur, 1=entire dur of note
          ifmdec        = p10*p3                ; % of total duration
          ifmoff        = p3 - (ifmrise + ifmdec)
          index         = p11
          ivibdepth     = p12
          ivibrate      = p13
          iformantamp   = p14                   ; % of total amp, 1=dB amp as in p4
          iformantrise  = p15*p3                ; % of total dur, 1=entire dur of note
          iformantdec   = p3 - iformantrise

          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)

          kfrmnt        linseg  0, iformantrise, iformantamp, iformantdec, 0
          kvib          oscil   ivibdepth,ivibrate,1
          afrmnt1       fof     iamp, p5+kvib, 650, 0, 40, .003, .017, .007, 4, 1, 2, p3
          afrmnt2       fof     iamp, (p5*1.001)+kvib*.009, 650, 0, 40, .003,.017,.007, 10,1,2,p3
          aformnt       =       kfrmnt * (afrmnt1+afrmnt2)

                        out     apluck + afm + aformnt
        endin

f1  0  8192  10  1                      ; sine wave
f2  0  2048   19  .5  1  270  1         ; sigmoid rise

;ins    st  dr   mp    frq   plkmp  plkdr  fmp   fmris  fmdec  indx  vbdp  vbrt  frmp fris
 i13    0   5    80    200   .8     .3     .7    .2     .35    8     1     5       3   .5
 i13    +   8    80    100   .      .4     .7    .35    .35    7     1     6       3   .7
 i13    .   13   80    50    .      .3     .7    .2     .4     6     1     4       3   .6
```

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:

1. You typed the letter **l** instead of the number **1**
2. You forgot to precede your comment with a semi-colon
3. You forgot an opcode or a required parameter
4. Your amplitudes are not loud enough or they are too loud
5. Your frequencies are not in the audio range - 20Hz to 20kHz
6. You placed the value of one parameter in the p-field of another
7. You left out some crucial information like a function definition
8. 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 this chapter, 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.

The directory 'morefiles' contains examples of the classical designs of Risset and Chowning. Detailed discussions of these instruments can be found in Charles Dodge's and Thomas Jerse's **Computer Music** textbook. This text is the key to getting the most out of these instrumental models and their innovative approaches to signal processing. Also recommended are the designs of Russell Pinkston. They demonstrate techniques for legato phrasing, portamento, random vibrato, and random sequence generation. His instrument representing Dx7 OpCode™ Editor/Librarian patches is a model for bringing many wonderful sounds into your orchestra.

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 - March 1991 - Boston, Massachusetts - USA

Appendix 3: An FOF Synthesis Tutorial

by

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The **fof** synthesis generator in Csound has more parameter fields than other modules. To help the user become familiar with these parameters this tutorial will take a simple orchestra file using just one **fof** unit-generator and demonstrate the effect of each parameter in turn. To produce a good vocal imitation, or a sound of similar sophistication, an orchestra containing five or more **fof** generators is required and other refinements (use of random variation of pitch etc.) must be made. The sounds produced in these initial explorations will be much simpler and consequently less interesting but they will help to show clearly the basic elements of **fof** synthesis. This tutorial assumes a basic working knowledge of Csound itself. The specification of the **fof** unit-generator (as found in the main Csound manual) is:

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

where	xamp, xfund, xform	can receive any rate (constant, control or audio)
	koct, kband, kdris, kdur, kdec	can receive only constants or control rates
	iolaps, ifna, ifnb, itotdur	must be given a fixed value at initialization
	[iphs][ifmode]	are optional, defaulting to 0.

The following orchestra contains a simple instrument we will use for exploring each parameter in turn. On the faster machines (DECstation, SparcStation, SGI Indigo) it will run in real time.

```
        sr = 22050
        kr = 441
        ksmps = 50
        instr   1
a1      fof     15000, 200, 650, 0, 40, .003, .02, .007, 5, 1, 2, p3
        out     a1
        endin
```

It should be run with the following score:

```
f1  0  4096  10  1
f2  0  1024  19  .5  .5  270  .5
i1  0  3
e
```

The result is very basic. This is not surprising since we have created only one formant region (a vocal imitation would need at least five) and have no vibrato or random variation of the parameters. By varying one parameter at a time we will help the reader learn how the unit-generator works. Each of the following "variations" starts from the model. Parameters not specified remain as given.

xamp = amplitude

The first input parameter controls the amplitude of the generator. At present our model uses a constant amplitude, this can be changed so that the amplitude varies according to a line function:

```
a2      linseg  0,  p3*.3,  20000,  p3*.4,  15000,  p3*.3,  0
a1      fof     a2, ......(as  before)...
```

The amplitude of a **fof** generator needs care. *xamp* does not necessarily indicate the maximum output, which can also depend on the rise pattern, bandwidth, and the presence of any "overlaps".

xfund = fundamental frequency

This parameter controls the pitch of the fundamental of the unit generator. Starting again from the original model this example demonstrates an exaggerated vibrato:

```
a2      oscil   20,  5,  1
a1      fof     15000,  200+a2,  etc........
```

fof synthesis produces a rapid succession of (normally) overlapping excitations or granules. The fundamental is in fact the speed at which new excitations are formed and if the fundamental is very low these excitations are heard as separate granules. In this case the fundamental is not so much a pitch as a pulse speed. The possibility of moving between pitch and pulse, between timbre and granular texture is one of the most interesting aspects of **fof**. For a simple demonstration try the following variation. It will be especially clear if the score note is lengthened to about 10 seconds.

```
a2      expseg  5,  p3*.8,  200,  p3*.2,  150
a1      fof      15000,  a2  etc........
```

koct = octaviation coefficient

Skipping a parameter, we come to an unusual means of controlling the fundamental: *octaviation*. This parameter is normally set to 0. For each unit increase in *koct* the fundamental pitch will drop by one octave. The change of pitch is **not** by the normal means of glissando, but by gradually fading out alternate excitations (leaving half the original number). Try the following (again with the longer note duration):

```
k1      linseg  0,  p3*.1,  0,  p3*.8,  6,  p3*.1,  6
a1      fof     15000,  200,  650,  k1  etc.........
```

This produces a drop of six octaves; if the note is sufficiently long you should be able to hear the fading out of alternate excitations towards the end.

xform = formant frequency; *ifmode* = formant mode (0 = striated, non-0 = smooth)

The spectral output of a **fof** unit-generator resembles that of an impulse generator filtered by a band pass filter. It is a set of partials above a fundamental *xfund* with a spectral peak at the formant frequency *xform*. Motion of the formant can be implemented in two ways. If *ifmode* = 0, data sent to *xform* has effect only at the start of a new excitation. That is, each excitation gets the current value of this parameter at the time of creation and holds it until the excitation ends. Successive overlapping excitations can have different formant frequencies, creating a richly varied sound. This is the mode of the original CHANT program. If *ifmode* is non-zero, the frequency of each excitation varies continuously with *xform*. This allows glissandi of the formant frequency. To demonstrate these differences we take a very low fundamental so that the granules can be heard separately and the formant frequency is audible not as the center frequency of a "band" but as a pitch in its own right. Compare the following in which only *ifmode* is changed:

```
a2      line    400,  p3,  800
a1      fof     15000,  5,  a2,  0,  1,  .003,  .5,  .1,  3,  1,  2,  p3,  0,  0
```

```
a2      line    400,  p3,  800
a1      fof     15000,  5,  a2,  0,  1,  .003,  .5,  .1,  3,  1,  2,  p3,  0,  1
```

In the first case the formant frequency moves by step at the start of each excitation, whereas in the second it changes smoothly. A more subtle difference is perceived with higher fundamental frequencies. (Note that the later **fof** parameters were changed in this example to lengthen the excitations so that their pitch could be heard easily.)

xform also permits frequency modulation of the formant frequency. Applying FM to an already complex sound can lead to strange results, but here is a simple example:

```
acarr   line    400,  p3,  800
index   =       2.0
imodfr  =       400
idev    =       index * imodfr
amodsig oscil   idev, imodfr, 1
a1      fof     15000,  5,  acarr+amodsig,  0,  1,  .003,  .5,  .1,  3,  1,  2,  p3,  0,  1
```

kband = formant bandwidth
kris, kdur, kdec = risetime, duration and decaytime (in seconds) of the excitation envelope

These parameters control the shape and length of the **fof** granules. They are shaped in three segments: a rise, a middle decay, and a terminating decay. For very low fundamentals these are perceived as an amplitude envelope, but with higher fundamentals (above 30 Hz) the granules merge together and these parameters effect the timbre of the sound. Note that these four parameters influence a new granule only at the time of its initialization and are fixed for its duration; later changes will affect only subsequent granules. We begin our examination with low frequencies.

```
k1      line    .003,  p3,  .1                                          ; kris
a1      fof     15000,  2,  300,  0,  0,  k1,  .5,  .1,  2,  1,  2,  p3
```

Run this with a note length of 10 seconds. Notice how the attack of the envelope of the granules lengthens. The shape of this attack is determined by the forward shape of *ifnb* (here a sigmoid).

Now try changing kband:

```
k1      linseg  0,  p3,  10                                             ; kband
a1      fof     15000,  2,  300,  0,  k1,  .003,  .5,  .1,  2,  1,  2,  p3
```

Following its rise, an excitation has a built-in exponential decay and *kband* determines its rate. The bigger *kband* the steeper the decay; zero means no decay. In the above example the successive granules had increasingly fast decays.

```
k1      linseg  .3,  p3,  .003
a1      fof     15000,  2,  300,  0,  0,  .003,  .4,  k1,  2,  1,  2,  p3
```

This demonstrates the operation of *kdec*. Because an exponential decay never reaches zero it must be terminated gracefully. *kdur* is the overall duration (in seconds from the start of the excitation), and *kdec* is the length of the terminating decay. In the above example the terminating decay starts very early in the first granules and then becomes progressively later. Note that *kband* is set to zero so that only the terminating decay is evident.

In the next example the start time of the termination remains constant, but its length gets shorter:

```
k1      expon  .3,  p3,  .003
a1      fof     15000,  2,  300,  0,  0,  .003,  .01 + k1,  k1,  2,  1,  2,  p3
```

It may be surprising to find that for higher fundamentals the local envelope determines the spectral shape of the sound. Electronic and computer music has often shown how features of music we normally consider independent (such as pitch, timbre, rhythm) are in fact different aspects of the same thing. In general, the longer the local envelope segment the narrower the band of partials around that frequency. *kband* determines the bandwidth of the formant region at -6dB, and *kris* controls the skirtwidth at -40dB. Increasing *kband* increases the local envelope's exponential decay rate, thus shortening it and increasing the -6dbB spectral region. Increasing *kris* (the envelope attack time) inversely makes the -40dB spectral region smaller.

The next example changes first the bandwidth then the skirtwidth. You should be able to hear the difference.

```
k1    linseg  100,  p3/4,  0,  p3/4,  100,  p3/2,  100                    ; kband
k2    linseg  .003,  p3/2,  .003,  p3/4,  .01,  p3/4,  .003              ; kris
a1    fof     15000,  100,  440,  0,  k1,  k2,  .02,  .007,  3,  1,  2,  p3
```

[In the first half of the note *kris* remains constant while *kband* broadens then narrows again. In the second half, *kband* is fixed while *kris* lengthens (narrowing the spectrum) then returns again .]

Note that *kdur* and *kdec* don't really shape the spectrum, they simply tidy up the decay so as to prevent unwanted discontinuities which would distort the sound. For vocal imitations these parameters are typically set at .017 and .007 and left unchanged. With high ("soprano") fundamentals it is possible to shorten these values and save computation time (reduce overlaps).

iolaps = number of overlap spaces

Granules are created at the rate of the fundamental frequency, and new granules are often created before earlier ones have finished, resulting in overlaps. The number of overlaps at any one time is given by *xfund * kdur*. For a typical bass note the calculation might be 200 * .018 = 3.6, and for a soprano note 660 * .015 = 9.9. **fof** needs at least this number (rounded up) of spaces in which to operate. The number can be over-estimated at no computation cost, and at only a small space cost. If there are insufficient overlap spaces during operation, the note will terminate.

ifna, *ifnb* = stored function tables

Identification numbers of two function tables (see the **fof** entry in the manual proper).

itotdur = total duration within which all granules in a note must be completed

So that incomplete granules are not cut off at the end of a note **fof** will not create new granules if they will not be completed by the time specified. Normally given the value "p3" (the note length), this parameter can be changed for special effect; **fof** will output zero after time *itotdur*.

iphs = initial phase (optional, defaulting to 0).

Specifies the initial phase of the fundamental. Normally zero, but giving different **fof** generators different initial phases can be helpful in avoiding "zeros" in the spectrum.

Appendix 4: Hetro - an Adsyn Analysis Program

HETRO

hetro - hetrodyne filter analysis module for the Csound **adsyn** unit generator

SYNOPSIS

hetro [flags] infilename outfilename

DESCRIPTION

hetro takes an input file of audio samples, decomposes it into component sinusoids, and outputs a description of the components in the form of breakpoint amplitude and frequency tracks. Analysis is conditioned by the control flags below. A space is optional between flag and value.

-s<srate> sampling rate of the audio input file. The default value is 20000.

-b<begin> beginning time (in seconds) of the audio segment to be analyzed. The default is 0.0

-d<duration> duration (in seconds) of the audio segment to be analyzed. The default of 0.0 means to the end of the file. Maximum length is 32.766 seconds.

-f<begfreq> estimated starting frequency of the fundamental, necessary to initialize the filter analysis. The default value is 100 (cps).

-h<partials> number of harmonic partials sought in the audio file. Default is 10. Maximum 50.

-M<maxamp> maximum amplitude summed across all concurrent tracks. The default is 32767.

-m<minamp> amplitude threshold below which a track is dormant and will not contribute to output summation. Typical values: 128 (48 db down from full scale), 64 (54 db down), 32 (60 db down), 0 (no thresholding). The default is 32.

-n<brkpts> initial number of analysis breakpoints in each amplitude and frequency track, prior to thresholding (**-m**) and standard data reduction. The initial points are spread evenly over the duration. The default is 256.

-l<cutfreq> substitute a 3rd order Butterworth low-pass filter with cutoff frequency *cutfreq* (in cps), in place of the default averaging comb filter. The default is 0 (don't use).

EXAMPLE

A typical call to **hetro** might be:

```
hetro   -s44100  -b.5  -d2.5  -h16  -M24000  audiofile.test  adsynfile7
```

This will analyze 2.5 seconds of a file "audiofile.test", recorded at 44100 Herz, beginning .5 seconds from the start, and place the result in a file "adsynfile7". We request just the first 16 harmonics of the sound, with 256 initial breakpoint values per amplitude or frequency track, and a peak summation amplitude of 24000. The fundamental is estimated to begin at 100 Hz. Amplitude thresholding is at 60 db down. The Butterworth LPF is not enabled.

FILE FORMAT

The output file contains time-sequenced amplitude and frequency values for each partial of an additive complex audio source. The information is in the form of breakpoints (time, value, time, value, ...) using 16-bit integers in the range 0 - 32767. Time is given in milliseconds, and frequency in Hertz (cps). The breakpoint data is exclusively non-negative, and the values -1 and -2 uniquely signify the start of new amplitude and frequency tracks. A track is terminated by the value 32767. Before being written out, each track is data-reduced by amplitude thresholding and breakpoint consolidation.

A component partial is defined by two breakpoint sets: an amplitude set, and a frequency set. Within a composite file these sets may appear in any order (amplitude, frequency, amplitude ....; or amplitude, amplitude..., then frequency, frequency,...). During **adsyn** resynthesis the sets are automatically paired (amplitude, frequency) from the order in which they were found. There should be an equal number of each.

A legal **adsyn** control file could have following format:

```
-1  time1  value1  ...   timeK  valueK   32767          ; amplitude breakpoints for partial 1
-2  time1  value1  ...   timeL   valueL   32767         ; frequency breakpoints for partial 1
-1  time1  value1  ...   timeM  valueM  32767           ; amplitude breakpoints for partial 2
-2  time1  value1  ...   timeN   valueN  32767          ; frequency breakpoints for partial 2
-2   time1   value1   ..........
-2   time1   value1   ..........                        ; pairable tracks for partials 3 and 4
-1   time1   value1   ..........
-1   time2   value1   ..........
```

Appendix 5: The LpcAnal Program

LPCANAL

lpcanal - linear predictive coding analysis module for the Csound **lp** generators

SYNOPSIS

lpcanal [flags] infilename outfilename

DESCRIPTION

lpcanal performs both lpc and pitch-tracking analysis on a soundfile of 16-bit audio samples, to produce a time-ordered sequence of *frames* of control information suitable for Csound resynthesis. Analysis is conditioned by the control flags below. A space is optional between the flag and its value.

-**b**<begin>	beginning time (in seconds) of the audio segment to be analyzed. The default is 0.0
-**d**<duration>	Duration (in seconds) of the audio segment to be analyzed. The default of 0.0 means to the end of the file.
-**p**<npoles>	Number of poles for analysis. The default is 34, which is also the maximum.
-**h**<hopsize>	Hop size (in samples) between frames of analysis. This determines the number of frames per second (srate / hopsize) in the output control file. The analysis framesize is hopsize * 2 samples. The default value is 200.
-**C**<string>	Text for the comments field of the lpfile header. The default is the null string.
-**P**<mincps>	Lowest frequency (in cps) of pitch tracking. -P0 means no pitch tracking.
-**Q**<maxcps>	Highest frequency (in cps) of pitch tracking. The narrower the pitch range, the more accurate the pitch estimate.
-**v**<verbosity>	Level of terminal information during analysis. 0 = none, 1 = verbose, 2 = debug. The default is 0.

"infilename" should be a monaural soundfile of 16-bit linear samples, currently with an IRCAM soundfile header. If the name does not begin with '.' or '/', the file is expected in the directory indicated by the environment variable SFDIR. Output is directed to "outfilename" in the current or specified directory.

EXAMPLE

The command

lpcanal -p26 -d2.5 -P100 -Q400 audiofile.test lpfil22

will analyze the first 2.5 seconds of file "audiofile.test" in SFDIR, producing srate/200 frames per second, each containing 26-pole filter coefficients and a pitch estimate between 100 and 400 Hertz. Output will be placed in "lpfil22" in the current directory.

FILE FORMAT

Output is a file comprised of an identifiable header plus a set of frames of floating point analysis data. Each frame contains four values of pitch and gain information, followed by *npoles* filter coefficients. The file is readable by Csound's **lpread**.

lpcanal is an extensive modification of Paul Lanksy's lpc analysis programs.

Appendix 6: The PvAnal Program

by

Dan Ellis

MIT Media Lab

PVANAL

pvanal - Fourier analysis module for the Csound **pvoc** unit generator

SYNTAX

pvanal [flags] infilename outfilename

DESCRIPTION

pvanal converts an audio file (time-domain) into a series of short-time Fourier transform (STFT) frames at regular points throughout the analysis file (a frequency-domain representation). The output file can be used by **pvoc** to generate audio fragments based on the original sample, with timescales and pitches arbitrarily and dynamically modified. Analysis is conditioned by the flags below. A space is optional between the flag and its argument.

-n<frmsiz> STFT frame size, the number of samples in each Fourier analysis frame. Must be a power of two, in the range 16 to 16384. For clean results, a frame must be larger than the longest pitch period of the sample. However, very long frames result in temporal “smearing” or reverberation. The bandwidth of each STFT bin is determined by sampling rate / frame size. The default framesize is the smallest power of two that corresponds to more than 20 milliseconds of the source (e.g. 256 points at 10 kHz sampling, giving a 25.6 ms frame).

-w<windfact> Window overlap factor. This controls the number of Fourier transform frames per second. Csound's **pvoc** will interpolate between frames, but too few frames will generate audible distortion; too many frames will result in a huge analysis file. A good compromise for *windfact* is 4, meaning that each input point occurs in 4 output windows, or conversely that the offset between successive STFT frames is framesize/4. The default value is 4. Do not use this flag with **-h**.

-h<hopsize> STFT frame offset. Converse of above, specifying the increment in samples between successive frames of analysis (see also lpcanal). Do not use with **-w**.

EXAMPLE

The command

pvanal asound.irc pvfile

will analyze the soundfile *asound.irc* using the default *frmsiz* and *windfact* to produce the file *pvfile* suitable for use with **pvoc**.

FILES

The output file has a special **pvoc** header containing details of the source audio file, the analysis frame rate and overlap. Frames of analysis data are stored as *float,* with the magnitude and 'frequency' (in Hz) for the first N/2 + 1 Fourier bins of each frame in turn. 'Frequency' encodes the phase increment in such a way that for strong harmonics it gives a good indication of the true frequency. For low amplitude or rapidly moving harmonics it is less meaningful.

DIAGNOSTICS

Prints total number of frames, and frames completed on every 20th.

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Appendix 7: Csound for the Macintosh

by

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Introduction

This document describes the Macintosh verson of the Csound program and assumes the reader is already familiar with the Csound program as described in the Csound Users Manual. Csound is primarily intended for the UNIX operating system and hence its operation is specified through command line arguments. The Macintosh version of Csound surrounds this mechanism with the standard Macintosh user interface primitives, e.g. menus and dialog boxes. After the user specifies the Csound input files and options using the Macintosh user interface, Macintosh Csound creates the UNIX command line and invokes Csound appropriately. All subsequent Csound output is directed to a console window (or optionally to a listing file). Output sound files are created in Digidesign's Sound Designer II format or optionally in AIFF format.

Orchestra and Score file selection dialog

When Csound is launched, it automatically brings up the main file selection dialog. This dialog has fields for the required input orchestra and score files, the output sample file, and optionally a MIDI file and output listing file. Only the output file name may be explicitly typed in, the other fields must be filled by clicking on the corresponding Select button, which will bring up a standard Macintosh file selection dialog. Because the orchestra and score file names usually differ only in the filename extension (".orc" for orchestra files and ".sco" for score files), Csound only requires that you select one of the two; the other file name will be automatically formed by changing the extension appropriately. If an output file is selected (in the Options dialog), then the output file name is similarly created with the extension ".snd".

Users familiar with Csound will recall that Csound expects all sound files to live in a single directory called the SFDir (for sound file directory). On UNIX, this directory is specified via a UNIX shell environment variable. On the Macintosh, the user must select this directory. This can be done by clicking on the SFDir button next to the output file text item. This brings up the Sound File Directory selection dialog which shows the current sound file directory. When shipped, this will be blank. **The sound file directory must be set up properly in order that output files may be created.** Clicking on the Select button brings up a standard file selection dialog. Navigate to the directory you want to use for sound files, and then click on the Save button. Finally, click on the Save Settings button so that Csound will remember this directory when it is run in the future.

If an input MIDI file is desired, click on the MIDI file checkbox. This brings up a file selection dialog which can be used to select a MIDI file.

If a listing file is desired, click on the listing file checkbox. This bring up a file selection dialog which can be used to specify an output listing file. If a listing file is selected, Csound will route almost all output to the listing file. Certain messages will still appear in the console window.

The Smp Fmt and Options buttons bring up other dialogs for setting the Csound output sample format and options, respectively. These are described later in this document.

After all the files and options have been set up, click on the OK button to run Csound. If the files and options are incorrectly set up, Csound will report an error by printing an appropriate message to the console window. After the score file has been processed, Csound will display the message

"*** PRESS MOUSE BUTTON TO EXIT ***". Pressing the mouse button will cause Csound to exit back to the Macintosh Finder. It is not possible to process multiple files without relaunching Csound.

Clicking on the Cancel button causes the file selection dialog to disappear. The dialog can be brought back by selecting the Choose Orchestra and Score menu item in the Csound menu. Note that cancelling the dialog does not cause Csound to forget the settings of the various options. This is particularly useful if you want to select extra options explicity by using the "Enter command line..." menu item.

Csound menu

This section describes the menu items available in the Csound menu. To access the menu items the file selection dialog must be cancelled as described above.

Choose Orchestra and Score...

Selecting this menu item brings up the main file selection dialog described above.

Options...

Selecting this menu item brings up the options dialog containing checkboxes for each Csound option. These are each described below:

Note amplitudes

When checked, causes information regarding note amplitudes to be displayed. Corresponds to the -m1 option in UNIX Csound.

Samples out of range

When checked, causes information regarding out of range samples to be displayed. Corresponds to the -m2 option in UNIX Csound.

Warnings

When checked, causes information regarding out of range samples to be displayed. Corresponds to the -m4 option in UNIX Csound.

No table graphics

When checked, suppresses the display of Csound wavetables. Corresponds to the -d option in UNIX Csound.

Diagnostic messages

When checked, causes diagnostic messages to be displayed. Corresponds to the -v option in UNIX Csound.

Initialize processing only

When checked, causes Csound to only perform initialization of orchestras and no other processing. Corresponds to the -I option in UNIX Csound.

No sound output

When checked, suppresses the output of a sound file. Corresponds to the -n option in UNIX Csound.

Change file types

When checked, Csound will change the file creator field of the selected orchestra and score files. This lets the Macintosh Finder know that the files are associated with the Csound application. Thus, if you subsequently double-click on one of these files from the Finder, Csound will be executed using the selected file as input.

Output Sample Format...

Selecting this menu item brings up the output sample format dialog which controls the format of the output sample file. All sound files are created as Digidesign Sound Designer II format files, unless the -A option is specified in the command line, which causes Csound to create AIFF (Audio Interchange File Format) files. The radio buttons select the sample format and default to 16-bit integer. The 8-bit integer, 8-bit a-law, 8-bit μ-law, 16-bit integer, 32-bit integer, and 32-bit float formats correspond to the -c, -a, -u, -s, -l, and -f options, respectively, in UNIX Csound. Note that 8-bit a-law format is not supported. If the no header checkbox is checked, this suppresses the output of a sound file header; only the raw samples are output. This corresponds to the -h option in UNIX Csound. The blocksize controls how many samples are accumulated before writing to the output sample file. This corresponds to the -b option in UNIX Csound. Note that all input files must be either Digidesign Sound Designer II format, AIFF format or raw 16-bit samples.

Sound File Directory...

Selecting this menu item brings up the sound file directory dialog. This dialog is described above in the file selection dialog section.

Sampled Sound Directory...

Selecting this menu item brings up the sampled sound directory dialog. This dialog allows the user to specify the directory where Csound will look for sampled sound files to be loaded into function tables. This corresponds to the SSDir shell environment variable in UNIX Csound. If no directory is specified, Csound will look for sampled sounds in the sound file directory, described earlier.

Enter Command Line...

Selecting this menu item brings up the command line dialog. The command line dialog is used for directly entering a UNIX command line to invoke Csound. This is useful for specifying obscure arguments to Csound which are not otherwise supported in the Macintosh interface. (The AIFF file option -A is one such option). The dialog comes up with a command line that corresponds to all currently selected files and options. The command line appears in a Macintosh text edit field for editing. When specifying file name arguments that contain spaces, enclose the argument in double-quotes. Click on OK to invoke Csound with the specified arguments, or click on Cancel to exit the dialog without executing Csound.

Save Settings

Selecting this menu item causes all option settings to be remembered for the next time Csound is executed.

Quit

Selecting this menu item causes Csound to exit to the Macintosh Finder.

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Appendix 8: Adding your own Cmodules to Csound

If the existing Csound generators do not suit your needs, you can write your own modules in **C** and add them to the run-time system. When you invoke Csound on an orchestra and score file, the orchestra is first read by a table-driven translator 'otran' and the instrument blocks converted to coded templates ready for loading into memory by 'oload' on request by the score reader. To use your own C-modules within a standard orchestra you need only add an entry in otran's table and relink Csound with your own code.

The translator, loader, and run-time monitor will treat your module just like any other provided you follow some conventions. You need a *structure* defining the inputs, outputs and workspace, plus some *initialization code* and some *perf-time code*. Let's put an example of these in two new files, **newgen.h** and **newgen.c**:

```
typedef struct {                                    /*  newgen.h  -  define a structure    */
        OPDS        h;                              /* required header       */
        float       *result, *istrt, *incr, *itime, *icontin; /* addr outarg, inargs  */
        float       curval, vincr;                  /* private dataspace     */
        long        countdown;                      /* ditto                 */
}  RMP;

#include "cs.h"                                     /*  newgen.c  -  init and perf code   */
#include "newgen.h"

void rampset(p)                                     /* at note initialization:        */
 register  RMP  *p;
{
        if  (*p->icontin == 0.)
                p->curval = *p->istrt;              /* optionally get new start value */
        p->vincr = *p->incr / esr;                  /* set s-rate increment per sec.  */
        p->countdown = *p->itime * esr;             /* counter for itime seconds      */
}

void ramp(p)                                        /* during note performance:    */
 register  RMP  *p;
{
        register float *rsltp = p->result;          /* init an output array pointer  */
        register int nn = ksmps;                    /* array size from orchestra     */
        do {
                *rsltp++ = p->curval;               /* copy current value to ouput */
                if (--p->countdown >= 0)            /* for the first itime seconds,  */
                        p->curval += p->vincr;      /*        ramp the value         */
        } while (--nn);
}
```

Now we add this module to the translator table **entry.c**, under the opcode name **rampt**:

```
#include "newgen.h"
void    rampset(), ramp();

/*  opcode      dspace        thread  outarg  inargs   isub      ksub     asub  */
{ "rampt",      S(RMP),       5,      "a",    "iiio",  rampset,  NULL,    ramp  },
```

Finally we relink Csound to include the new module. Under Unix this means changing the Makefile in three places:

1. Add the name newgen.o to the variable OBJS.
2. Add the name newgen.h as a dependency for entry.o
3. Create a new dependency, newgen.o: newgen.h

Now run 'make csound'. If your host is a Macintosh, simply add newgen.h and newgen.c to one of the Csound segments and invoke the C compiler.

The above actions have added a new generator to the Csound language. It is an audio-rate linear ramp function which modifies an input value at a user-defined slope for some period. A ramp can optionally continue from the previous note's last value. The Csound manual entry would look like:

ar **rampt** istart, islope, itime [, icontin]

istart - beginning value of an audio-rate linear ramp. Optionally overridden by a *continue* flag.

islope - slope of ramp, expressed as the y-interval change per second.

itime - ramp time in seconds, after which the value is held for the remainder of the note.

icontin (optional) - continue flag. If zero, ramping will proceed from input *istart* . If non-zero, ramping will proceed from the last value of the previous note. The default value is zero.

The file *newgen.h* includes a one-line list of output and input parameters. These are the ports through which the new generator will communicate with the other generators in an instrument. Communication is by *address*, not *value*, and this is a list of pointers to floats. There are no restrictions on names, but the input-output argument types are further defined by character strings in entry.c (inargs, outargs). Inarg types are commonly **x**, **a**, **k**, and **i**, in the normal Csound manual conventions; also available are **o** (optional, defaulting to 0), **p** (optional, defaulting to 1). Outarg types include **a**, **k**, **i** and **s** (asig or ksig). It is important that all listed argument names be assigned a corresponding argument type in entry.c. Also, i-type args are valid only at initialization time, and other-type args are available only at perf time. Subsequent lines in the RMP structure declare the work space needed to keep the code re-entrant. These enable the module to be used multiple times in multiple instrument copies while preserving all data.

The file *newgen.c* contains two subroutines, each called with a pointer to the uniquely allocated RMP structure and its data. The subroutines can be of three types: note initialization, k-rate signal generation, a-rate signal generation. A module normally requires two of these—initialization, and either k-rate or a-rate subroutines—which become inserted in various threaded lists of runnable tasks when an instrument is activated. The thread-types appear in entry.c in two forms: *isub*, *ksub* and *asub* names; and a threading index which is the sum of isub=1, ksub=2, asub=4. The code itself may reference global variables defined in **cs.h** and **oload.c**, the most useful of which are:

float	esr	user-defined sampling rate
float	ekr	user-defined control rate
float	ensmps	user-defined ksmps
int	ksmps	user-defined ksmps
int	nchnls	user-defined nchnls
int	odebug	command-line -v flag
int	initonly	command-line -I flag
int	msglevel	command-line -m level
float	pi, twopi	obvious constants
float	tpidsr	twopi / esr
float	sstrcod	special code for string arguments

Function tables

To access stored function tables, special help is available. The newly defined structure should include a pointer

```
FUNC        *ftp;
```

initialized by the statement

```
ftp = ftpfind(p->ifuncno);
```

where float *ifuncno is an i-type input argument containing the ftable number. The stored table is then at ftp->ftable, and other data such as length, phase masks, cps-to-incr converters, are also accessed from this pointer. See the FUNC structure in cs.h, the ftfind() code in fgens.c, and the code for oscset() and koscil() in ugens2.c.

Additional space

Sometimes the space requirement of a module is too large to be part of a structure (upper limit 65535 bytes), or it is dependent on an i-arg value which is not known until initialization. Additional space can be dynamically allocated and properly managed by including the line

```
AUXCH      auxch;
```

in the defined structure (*p), then using the following style of code in the init module:

```
if (p->auxch.auxp == NULL)
        auxalloc(npoints * sizeof(float),  &p->auxch);
```

The address of this auxilliary space is kept in a chain of such spaces belonging to this instrument, and is automatically managed while the instrument is being duplicated or garbage-collected during performance. The assignment

```
char *auxp = p->auxch.auxp;
```

will find the allocated space for init-time and perf-time use. See the LINSEG structure in ugens1.h and the code for lsgset() and klnseg() in ugens1.c.

File sharing

When accessing an external file often, or doing it from multiple places, it is often efficient to read the entire file into memory. This is accomplished by including the line

```
MEMFIL        *mfp;
```

in the defined structure (*p), then using the following style of code in the init module:

```
if (p->mfp == NULL)
        p->mfp = ldmemfile(filname);
```

where char *filname is a string name of the file requested. The data read will be found between

(char *) p->mfp->beginp;   *and*   (char *) p->mfp->endp;

Loaded files do not belong to a particular instrument, but are automatically shared for multiple access. See the ADSYN structure in ugens3.h and the code for adset() and adsyn() in ugens3.c.

String arguments

To permit a quoted string input argument (float *ifilnam, say) in our defined structure (*p), assign it the argtype **S** in entry.c, include another member char *strarg in the structure, insert a line

```
TSTRARG( "rampt",  RMP)  \
```

in the file **oload.h**, and include the following code in the init module:

```
if (*p->ifilnam == sstrcod)
        strcpy(filename, unquote(p->strarg));
```

See the code for adset() in ugens3.c, lprdset() in ugens5.c, and pvset() in ugens8.c.

Appendix 9: A CSOUND QUICK REFERENCE

VALUE CONVERTERS

ftlen(x)	(init-rate args only)
int(x)	(init- or control-rate args only)
frac(x)	" "
dbamp(x)	" "
i(x)	(control-rate arg; only)
abs(x)	(no rate restriction)
exp(x)	" "
log(x)	" "
sqrt(x)	" "
sin(x)	" "
cos(x)	" "
ampdb(x)	" "

PITCH CONVERTERS

octpch(pch)	(init- or control-rate args only)
pchoct(oct)	" "
cpspch(pch)	" "
octcps(cps)	" "
cpsoct(oct)	(no rate restriction)

PROGRAM CONTROL

igoto	label
tigoto	label
kgoto	label
goto	label
if	ia R ib **igoto** label
if	ka R kb **kgoto** label
if	ia R ib **goto** label
timout	istrt, idur, label

MIDI CONVERTERS

iamp	**ampmidi**	iscal[, ifn]
kaft	**aftouch**	iscal
kchpr	**chpress**	iscal
kbend	**pchbend**	iscal
ival	**midictrl**	inum
kval	**midictrl**	inum

SIGNAL GENERATORS

kr	**line**	ia, idur1, ib
ar	**line**	ia, idur1, ib
kr	**expon**	ia, idur1, ib
ar	**expon**	ia, idur1, ib
kr	**linseg**	ia, idur1, ib[, idur2, ic[...]]
ar	**linseg**	ia, idur1, ib[, idur2, ic[...]]
kr	**expseg**	ia, idur1, ib[, idur2, ic[...]]
ar	**expseg**	ia, idur1, ib[, idur2, ic[...]]
kr	**phasor**	kcps[, iphs]
ar	**phasor**	xcps[, iphs]
ir	**table**	indx, ifn[, ixmode][, ixoff][, iwrap]
ir	**tablei**	indx, ifn[, ixmode][, ixoff][, iwrap]
kr	**table**	kndx, ifn[, ixmode][, ixoff][, iwrap]
kr	**tablei**	kndx, ifn[, ixmode][, ixoff][, iwrap]
ar	**table**	andx, ifn[, ixmode][, ixoff][, iwrap]
ar	**tablei**	andx, ifn[, ixmode][, ixoff][, iwrap]
kr	**oscil1**	idel, kamp, idur, ifn
kr	**oscil1i**	idel, kamp, idur, ifn
kr	**oscil**	kamp, kcps, ifn[, iphs]
kr	**oscili**	kamp, kcps, ifn[, iphs]
ar	**oscil**	xamp, xcps, ifn[, iphs]
ar	**oscili**	xamp, xcps, ifn[, iphs]
ar	**foscil**	xamp, kcps, kcar, kmod, kndx, ifn[, iphs]
ar	**foscili**	xamp, kcps, kcar, kmod, kndx, ifn[, iphs]
ar1 [,ar2]	**loscil**	xamp, kcps, ifn[, ibas][, imod1,ibeg1,iend1] [, imod2,ibeg2,iend2]
ar	**buzz**	xamp, xcps, knh, ifn[, iphs]
ar	**gbuzz**	xamp, xcps, knh, kih, kr, ifn[, iphs]
ar	**adsyn**	kamod, kfmod, ksmod, ifilcod
ar	**pvoc**	ktimpnt, kfmod, ifilcod[, ispecwp]
ar	**fof**	xamp, xfund, xform, koct, kband, kris, kdur, kdec, iolaps, ifna, ifnb, itotdur[, iphs][, ifmode]
ar	**pluck**	kamp, kcps, icps, ifn, imeth [, iparm1, iparm2]
kr	**rand**	xamp[, iseed]
kr	**randh**	kamp, kcps[, iseed]
kr	**randi**	kamp, kcps[, iseed]
ar	**rand**	xamp[, iseed]
ar	**randh**	xamp, xcps[, iseed]
ar	**randi**	xamp, xcps[, iseed]

SIGNAL MODIFIERS

kr	**linen**	kamp, irise, idur, idec
ar	**linen**	xamp, irise, idur, idec
kr	**linenr**	kamp, irise, idec, iatdec
ar	**linenr**	xamp, irise, idec, iatdec
kr	**envlpx**	kamp, irise, idur, idec, ifn, iatss, iatdec[, ixmod]
ar	**envlpx**	xamp, irise, idur, idec, ifn, iatss, iatdec[, ixmod]
kr	**port**	ksig, ihtim[, isig]
ar	**tone**	asig, khp[, istor]
ar	**atone**	asig, khp[, istor]
ar	**reson**	asig, kcf, kbw[, iscl, istor]
ar	**areson**	asig, kcf, kbw[, iscl, istor]
krmsr,krmso,kerr,kcps	**lpread**	ktimpnt, ifilcod[, inpoles][, ifrmrate]
ar	**lpreson**	asig
ar	**lpfreson**	asig, kfrqratio
kr	**rms**	asig[, ihp, istor]
nr	**gain**	asig, krms[, ihp, istor]
ar	**balance**	asig, acomp[, ihp, istor]
kr	**downsamp**	asig[, iwlen]
ar	**upsamp**	ksig
ar	**interp**	ksig[, istor]
kr	**integ**	ksig[, istor]
ar	**integ**	asig[, istor]
kr	**diff**	ksig[, istor]
ar	**diff**	asig[, istor]
kr	**samphold**	xsig, kgate[, ival, ivstor]
ar	**samphold**	asig, xgate[, ival, ivstor]
ar	**delayr**	idlt[, istor]
	delayw	asig
ar	**delay**	asig, idlt[, istor]
ar	**delay1**	asig[, istor]
ar	**deltap**	kdlt
ar	**deltapi**	xdlt
ar	**comb**	asig, krvt, ilpt[, istor]
ar	**alpass**	asig, krvt, ilpt[, istor]
ar	**reverb**	asig, krvt[, istor]

OPERATIONS USING SPECTRAL DATA TYPES

dsig	**octdown**	xsig, iocts, isamps[, idisprd]
wsig	**noctdft**	dsig, iprd, ifrqs, iq[, ihann, idbout, idsines]
wsig	**specscal**	wsigin, ifscale, ifthresh
wsig	**specaddm**	wsig1, wsig2[, imul2]
wsig	**specdiff**	wsigin
wsig	**specaccm**	wsigin
wsig	**specfilt**	wsigin, ifhtim
	specdisp	wsig, iprd[, iwtflg]
ksum	**specsum**	wsig[, interp]

SENSING & CONTROL

ktemp	**tempest**	kin, iprd, imindur, imemdur, ihp, ithresh, ihtim, ixfdbak, istartempo, ifn[, idisprd, itweek]
kx, ky	**xyin**	iprd, ixmin, ixmax, iymin, iymax[, ixinit, iyinit]
	tempo	ktempo, istartempo

SOUND INPUT & OUTPUT

a1	**in**	
a1, a2	**ins**	
a1, a2, a3, a4	**inq**	
a1	**soundin**	ifilcod[, iskptim][, iformat]
a1, a2	**soundin**	ifilcod[, iskptim][, iformat]
a1, a2, a3, a4	**soundin**	ifilcod[, iskptim][, iformat]
	out	asig
	outs1	asig
	outs2	asig
	outs	asig1, asig2
	outq1	asig
	outq2	asig
	outq3	asig
	outq4	asig
	outq	asig1, asig2, asig3, asig4
a1, a2, a3, a4	**pan**	asig, kx, ky, ifn[, imode][, ioffset]

SIGNAL DISPLAY

	print	iarg[, iarg,...]
	display	xsig, iprd[, iwtflg]
	dispfft	xsig, iprd, iwsiz[, iwtyp][, idbouti][, iwtflg]