

# *Impulse Tracker*

## User's Manual

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## **1. Introduction**

If you are familiar with tracker programs, then you could probably skip through most of this document. (I strongly suggest that ANYONE print out SUMMARY.TXT though, for a list of effects and editing commands.) For those of you who have had experience with *Scream Tracker 3*, read section **2.7**. This should detail most of the major differences between *Scream Tracker 3* and *Impulse Tracker*. For those of you who are unfamiliar with tracker programs it would probably be best to print out this document, and then refer to it as you work with the program.

Check out the file UPDATE.TXT. If this document contains conflicting information with what's listed in UPDATE.TXT, then follow what UPDATE.TXT contains, as I may have forgotten to update this text.

### **1.1 What is Impulse Tracker?**

*Impulse Tracker* is a program used to create high quality music without the requirements of specialised, expensive equipment. The hardware requirements should be easily met: Any IBM 386+ compatible computer (although a 486+ is recommended), and a VGA or higher video card. If you want to hear sound, you'll need one of the following:

- ⊗ Sound Blaster series (or most compatibles)

- ⊗ Pro Audio Spectrum
- ⊗ Windows Sound System
- ⊗ ESS ES1868 AudioDrive
- ⊗ Gravis UltraSound
- ⊗ Interwave based board
- ⊗ DAC on LPT1/LPT2
- ⊗ PC Speaker

Note that stereo effects (including surround sound) can *only* be experienced on software mixed stereo cards. Hardware mixed soundcards support panning, but do not support surround sound.

You'll need about 500k of conventional memory to get the program running - about 600k to have it load almost any song (as long as you have an EMM driver - samples are stored in EMS when possible, otherwise the songs that you are able to load will be limited even further!).

If you use EMM386.EXE, add "H=255" at the end of it to prevent "Out of memory" messages when you still have EMS remaining. If you're using some other memory manager, check it's documentation to see how to increase the number of "memory handles".

*Impulse Tracker* supports direct control over 64 channels, and can load and play the following formats: S3M, MTM, MOD and of course, IT. At the moment, modules can only be saved in the IT and S3M formats.

The number of channels playable on hardware mixed soundcards is limited by the hardware. Check DRIVERS.TXT for specific details on your soundcard hardware.

The *Gravis UltraSound* can only play 32 channels, although this shouldn't be too much of a limitation. Note that if you put notes in channels 33->64 in sample mode using a *Gravis*, you WILL NOT hear these notes! (There's an explanation of sample mode much later on, just keep it in mind if you own a *Gravis UltraSound* - it shouldn't cause any problems.) The same restriction applies for the *Sound Blaster AWE 32*, but to 30 channels. (ie. notes in channels 31->64 will not cause any sound)

## 1.2 About Impulse Tracker.

*Impulse Tracker* began simply as an extension to *Scream Tracker 3* (which should be quite obvious to *Scream Tracker 3* users, due to the interface.) At first, I only intended a couple of extra features (eg. proper panning and a couple of other interesting functions), but with the release of *Fast Tracker 2*, it became obvious that there were many areas in which *Scream Tracker* could be improved.

In spite of all the limitations of *Scream Tracker 3* (in comparison to *Fast Tracker 2*, which offered Volume/Panning Envelopes, 16-bit samples, Samples > 64k, an in built sampler, proper stereo panning on *SB16*, etc. etc), there were still more .S3M releases than there were .XM releases! And the reason must have been within the way you WRITE music with *Scream Tracker 3*. That's why I've used the same simple and QUICK interface that *Scream Tracker 3* offered. And I've also Incorporated all the power of *Fast Tracker 2* and more!

The Tracker was written in **100%** Assembler. All the routines are of my own coding (That's why some of them suck so much :)). The program was written on and off during 1995, but most of the work was completed in the summer holidays of '96. The source code is over 100,000 lines long and occupies over 3MB. The Tracker runs entirely in text mode (!) with some neat remapping of characters (that's why I haven't been able to use colours to help in some places ie. making the volume envelope nodes a different colour would have made it easier to use!).

## 1.3 Running Impulse Tracker.

*Win95* Users - ignore all this HIMEM and EMM386 stuff.. as long as you don't have "noems" or "noframe" as a parameter to EMM386 in your config.sys file, it'll work fine. If you have no idea what I'm talking about, you can probably assume it's fine :) (unless you get "out of memory messages")

*Impulse Tracker* uses EMS. If you want to be able to load large songs, you need the following lines in your CONFIG.SYS file (on your boot drive)

```
DEVICE=<path>\HIMEM.SYS
DEVICE=<path>\EMM386.EXE RAM H=255
```

eg. if the files are in your C:\DOS directory, you need this:

```
DEVICE=C:\DOS\HIMEM.SYS
DEVICE=C:\DOS\EMM386.EXE RAM H=255
```

If you use QEMM, use the following line instead:

```
DEVICE=<path>\QEMM.SYS DMA=64, HANDLES=255
```

If you want to run *Impulse Tracker* in *Windows 95*, check out WIN95.TXT

To find out details about the driver for your soundcard, check through DRIVERS.TXT.

Most users should not require any command line switches, however the following are available:

**S**filename.Drv - Sets sound driver to use. eg. IT /sITIW.DRV this may become necessary as new sound drivers are released.

**s#** Set sound card

- s0 = No sound card (silent mode)
- s1 = PC Speaker
- s2 = Sound Blaster
- s3 = Sound Blaster 2
- s4 = Sound Blaster Pro
- s5 = Sound Blaster 16
- s6 = Sound Blaster AWE 32
- s7 = Gravis UltraSound
- s8 = Interwave
- s9 = Pro Audio Spectrum
- s10 = Pro Audio Spectrum 16
- s11 = Windows Sound System
- s20 = .WAV writer device

If this parameter is omitted, then a (hardware) detection routine is used. Check DRIVERS.TXT for specific information on the various drivers.

**Axxx** Set Base Address of sound card (hex)  
**D#** Set DMA of sound card (decimal)  
**I##** Set IRQ of sound card (decimal)

**M#####** Set Mixspeed.

Values are accepted between 0->65535, however, the soundcards have hardware limits which will override the command line switch. Again, check DRIVERS.TXT for specific information on your sound driver.

**L###** Limit number of active channels

When you limit the number of channels, you are limiting the number of notes that you can hear on playback. In a "Sample"controlled song (explained later), the result is that if you play a 16 channel piece with the limit at 4 channels, then you will lose the last 12 channels! Even if the first 4 are not playing anything. In an instrument controlled song, the result is that you will hear 4 notes at

most, which can be controlled from any of the 64 channels, excess notes will be lost. (The reason for this is within the channel allocation routines.)

Note that the hardware mixed devices (eg. *Gravis UltraSound*, *AMD Interwave* and *Sound Blaster AWE32*) cannot play more than a certain number of channels (drivers.TXT will provide the exact details).

If you are new to tracking and didn't understand what I was just talking about, highlight this action with a marker (you've printed this out, right?), and in your dabbings, if you notice that some notes aren't being played, reread this section. It should make sense by then, and the reason hopefully lies in the above paragraphs.

**v1** Override VGA detection routine.. in case you DO have a VGA and the program doesn't recognise it. If characters look weird, you may need to use this switch to override the Matrox mode autodetect.

**v2** Force Matrox mode.  
Use this switch if you get garbled stuff appearing on your screen

**R** Reverse stereo channels.  
This is only really useful on the *SB16*. (Swaps left/right outputs). The *Sound Blaster Pro* stereo setting routines aren't accurate, and will sometimes set the left to left and right to right (how it's supposed to be), and sometimes set the left to right and right to left (how it's not supposed to be :). You can also switch left/right channels in IT by pressing Alt-R on the info page.

**c** Control playback in DOS Shell.  
When this option is on, the following keys will operate within the DOS Shell:

**Right-Alt:** Stop playback  
**Right-Ctrl:** Play song (if not already playing)  
**Grey Plus:** Increase global volume  
**Grey Minus:** Decrease global volume

**Warning:** There are problems on some computers with this enabled! If your keyboard locks up, it may be necessary to press the left ctrl/alt to "unfreeze" it... sometimes (other times.. I don't know!)

**F** Disable file colour distinctions.  
When this is on, all file colours will appear the same as the background colour. This is just to make it easier to redo the palette to your liking more easily.

## 1.4 Technical Information

Formats supported

### Modules:

**MOD** (M.K., M!K!, 4CHN, 6CHN, 8CHN, xxCH, FLT4, FLT8)  
**669** (Composer 669, Unis669)  
**MTM** (MMEdit files)  
**S3M** (Scream Tracker 3 files)  
**XM** (Fast Tracker 2 files, DigiTracker 3 files)  
**IT** (Impulse Tracker 1.xx, Impulse Tracker 2.xx files)

### Samples:

**S3I** (Scream tracker sample, 8 bit or 16 bit)  
**IFF** (Fast tracker 2 sample, 8 bit or 16 bit)  
**WAV** (Microsoft WAV, 8 bit or 16 bit)  
**Wxx** (TX Wave format samples)

**ITS** (Impulse tracker sample, 8 bit or 16 bit)  
**RAW** (Raw sample information, assumed 8 bit)  
**PAT** (Gravis UltraSound patches)  
**KRZ** (Kurzweil Synth files)  
**MOD** (Samples from almost all .MOD formats)  
**PTM** (Samples from Poly Tracker modules)  
**669** (Samples from 669 and 669 Enhanced modules)  
**FAR** (Samples from Farandole composer modules)  
**MTM** (Samples from MMEdit modules)  
**S3M** (Samples from Scream Tracker 3 modules)  
**XM** (Samples from Fast Tracker 2 modules)  
**IT** (Samples from Impulse Tracker 1.xx, 2.xx modules)

**Instruments:**

**XI** (Instruments from Fast Tracker 2)  
**ITI** (Instruments from Impulse Tracker)  
**XM** (Instruments from Fast Tracker 2 modules)  
**IT** (Instruments from Impulse Tracker 1.xx, 2.xx modules)

***Modules***

99 Samples maximum  
99 Instruments maximum  
200 Patterns maximum (from 32-200 rows per pattern)  
256 Orders maximum  
64 Channels under direct control  
256 Channels maximum for virtual control (64 max default)

***Samples***

8/16 bit samples  
Maximum size around 4MB  
No Loop/Forwards/Ping pong loop  
Default pan for samples (optional)  
Vibrato parameters

***Instruments***

25-point Volume/Panning/Pitch envelopes  
Default pan for instruments (optional)  
Pitch pan separation  
Volume / panning swing  
Virtual controls (\*REALLY\* worth understanding)

***Sound Devices***

Sound Blaster 1.xx, 2.xx (8 bit mono)  
Sound Blaster Pro (8 bit stereo)  
Sound Blaster 16 (16 bit stereo, MIDI supported)  
Sound Blaster AWE 32 (16 bit stereo, hardware mixing, MIDI supported)  
Pro Audio Spectrum (8 bit stereo)  
Pro Audio Spectrum 16 (16 bit stereo)  
Windows Sound System card (16 bit stereo)  
ESS ES1868 AudioDrive (16 bit stereo)  
Gravis UltraSound (16 bit stereo, hardware mixing)  
Interwave based board (eg. GUS PnP, Dynasonix, 16 bit stereo, hardware mixing, MIDI supported)  
DAC on LPT1/LPT2 (8 bit mono)  
PC Speaker (very final option not recommended... )

## 2. Using Impulse Tracker

Before any specific details of how to use the tracker, it may be worth knowing these few "editing" keys:

*When using Thumbbars.*

Pressing **Left** and **Right** arrows will shift the values left/right Holding down **Ctrl** while pressing **Left/Right** will move them quicker And **Shift-Left/Right** will move them even quicker.... But just typing in a number will get you directly to the desired value!

*String (text) entry.*

Nothing special here (and the routines need to be rewritten), but pressing **Ctrl-Backspace** will clear the entire text..

*Numerical entries (of the 7 digit or 3 digit variety)*

Pressing '+' or '-' will increase/decrease the value by 1. This is especially useful when modifying loop values.

### 2.1 Playing songs

If you are like most of us, you would have run the program first, and probably figured out how to do this yourself. Songs are simply loaded by pressing **F9** at any time (to invoke the file load menu) and then selecting a song by pressing **Enter**. You can also type the first few characters of a filename, and a "search" will be made for the first match.

Note that on all the file menus in *Impulse Tracker*, you can also delete files by pressing **Delete**.

After loading a song, use **F5** to play it and **F8** to stop. The **Info Page** should appear, and you can cycle through views using **PgUp/PgDn**. Separate view windows can be created by pressing **Insert** (and removed by pressing delete) and these separate windows can each have a different view method.

Details (ranges follow in brackets):

<i>Frequency</i>	- the speed (pitch) at which the sample is played in samples per second
<i>Position</i>	- the offset in a sample (in bytes)
<i>Smp</i>	- The sample currently being played (1->99)
<i>FVL</i>	- The final volume of the sample, taking into account all the scaling factors (viz global volume, sample vol, envelope volume, channel volume and fadeout) (0->128)
<i>CV</i>	- Channel volume (0->64)
<i>SV</i>	- Sample volume (0->64)
<i>VE</i>	- Envelope volume (0->64)
<i>Fde</i>	- Fadeout component (0->512)
<i>Pn</i>	- Panning (0->64, Su = surround)
<i>PE</i>	- Panning envelope value (0->32)
<i>NNA</i>	- The current NNA (Cut/Con/Off/Fde)
<i>Tot</i>	- The total number of active virtual channels 'owned' by the channel

If you're looking for some songs to play, check out the following places:

<ftp.cdrom.com/pub/demos/music> [huge collection!]  
<kosmic.wit.com/kosmic/songs>  
<ftp.uni-muenster.de/pub/sounds>  
<archie.au/pub/aminet/mods>  
for .IT files, go to <ftp.cdrom.com/pub/demos/music/songs/1996/it>

Let me know of any other good music module distribution sites, so that I can include them here!

### 2.2 Pattern editor (F2)

The pattern editor allows you to edit patterns. For those of you who are familiar with music, patterns can be thought of as "bars", and the order in which these "bars" are played is determined by order list. For those of you



```

|C-4 01 .. ...| - will play note C, octave 4, instrument 1
|D-4 .. .. ...| - will play note D, octave 4, instrument 1
|... 02 .. ...| - Will play note D, octave 4, instrument 2
|E-4 .. .. ...| - will play note E, octave 4, instrument 2
|G-6 12 .. ...| - will play note G, octave 6, instrument 12

```

2) The second column contains the sample/instrument, depending on whether the tracker is operating in sample mode, or instrument mode. Valid ranges are from 01->99 (decimal).

3) The third column contains the volume or panning of the note. To toggle between entry of volumes and panning, press ``

Volumes.

0 is the softest (ie. nothing) and 64 (decimal!) is the loudest.

If no volume is specified, then the default volume for the sample is used. (Note that effects \*CANNOT\* raise the volume above 64)

The volume scale works linearly - ie. one note played at a volume of 64 will be the same 'loudness' as 4 of the same note, one played at a volume of 10, another at a volume of 30, another at a volume of 20 and another at a volume of 4. (10+30+20+4 = 64) But if you have a choice, always go for the single note (it will require less processing (occupy less channels), and also sound better in quality, due to roundings that occur in processing softer volumes).

### ***Panning.***

Panning controls appear in a different colour to the volume controls. For panning in this column, 0 represents far left, and 64 represents far right. If you have an Xxx effect at the same time, the Xxx will take precedence over this column. It is more efficient filesize-wise to use a panning control in this column rather than in the effect column.

### ***Volume, Panning & Effect??***

OK. So you want to do something tricky. Well, it's possible to get all 3 of these how you want with a little meddling with the channel volume control - just place one before the row with an appropriate value (remember that the range is 0->40 HEX) such that the channel volume scales the default volume to the desired value.... What I mean is that if you want to play a note at volume 32, pan 48, with vibrato H81, then you can do the following:

```

|... .. .. M20| - Set channel volume to HALF.
|C-4 01 48 H81| - The 48 is a pan command - this assumes that
                  the default volume for sample/instrument 1
                  is 64.

```

### ***Volume Column Effects***

In IT208 and higher, some extra functions are available in the volume column. These allow you to slide the volume up/down and pitch up/down, just like the final column effects. If you aren't familiar with the rest of the editor, leave this section out for now and come back after you have become acquainted to column (4) of the editor - the effects column.

Volume column effects are selected by pressing A-H in the first column of the effects.

```

Ax      = fine volume slide up by x
Bx      = fine volume slide down by x
Cx      = volume slide up by x
Dx      = volume slide down by x
Ex      = pitch slide down by x
Fx      = pitch slide up by x
Gx      = portamento to note with speed x
Hx      = vibrato with depth x

```



In all cases, if x is 0, then the effect memory is used (as explained in (4))

The memory for Ax/Bx/Cx/Dx are shared, as is the memory for Ex/Fx.

4) The final column contains effect data. For those of you who are just starting, I would advise you to leave this section until later, when you have already dabbled with entering notes, and want some special features. It's easy to get caught up with all these special features and you can 'overdo' the effects. And it will sound absolutely pathetic.

Trust me, I know - I've done it :)

Effects are entered by typing 'a'-z', then a hex value (see later for an explanation of hex numbers) as the final two entries. The effects allow you to a wide variety of functions that are otherwise impossible to obtain.

Due to the repetitive nature of some effects, there is a 'memory' so that instead of typing:

```
... .. G12   It is easier to   ... .. G12
... .. G12   use:                ... .. G00
... .. G12   ... .. G00
... .. G12   ... .. G00
... .. G12   ... .. G00
... .. G12   ... .. G00
... .. G12   ... .. G00
```

The following effects 'memorise' their previous values:  
(D/K/L), (E/F/G), (HU), I, J, N, O, S, T, W

Note: Bracketed commands share the same 'memory' value. So

```
... .. E12   can be written as:  ... .. E12
... .. F12   ... .. F00
... .. E12   ... .. E00
... .. F12   ... .. F00
C-4 01 .. G12   C-4 01 .. G00
```

Commands H and U are linked even more closely.

If you use H00 or U00, then the previous vibrato, no matter whether it was set with Hxx or Uxx will be used. So:

```
... .. H81   Is the same as:    ... .. H81
... .. U00   ... .. H81
... .. U83   ... .. U83
... .. U00   ... .. U83
... .. H00   ... .. U83
```

### **Hex Numbers**

Note: Impulse Tracker works ENTIRELY with decimal numbers EXCEPT for the effects column.

Instead of using a decimal system (ie. base 10), it is more natural for the computer to work with hexadecimal (often abbreviated to simply 'Hex') - numbers which operate in base 16. The first 9 numbers in hex are denoted by '1' to '9' and the next 6 are denoted by 'A' to 'F'. So if you count in hex, it will be as follows: (0), 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, 10, 11, 12, 13, 14, 15, 16, 17, 18, 19, 1A, 1B, 1C, 1D, 1E, 1F, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 2A etc.

To convert a hex number to decimal, multiply the 'tens' column by 16 and add the value of the second column. ie. 32 Hex = 3\*16+2 = 50 decimal. 2A hex = 2\*16+10 = 42. (because A = 10) The maximum number that you can represent with two Hex digits is FF = 255 decimal.

Hope that this makes SOME sense :)

## *Effects.*

### **Axx** *Set Speed.*

I prefer to think of this command as "Set Frames per Row". Normally, the tracker operates at around 50 frames a second. If the rows were played at this speed, then a huge amount of space would be required to enter the pattern data. Instead, setting the 'speed' of the song will cause the tracker to wait on the current row for 'xx' frames. Hence, setting the speed at 50 (decimal = 32hex) will cause each row to last about a second - quite a long time! The default is A06. The initial speed can be set in the variables screen on **F12**.

If two Axx commands are given in the same row, then the command in the higher channel (by number) will take effect.

### **Bxx** *Jump to order.*

Causes the song to jump to order xx (hex). This is often used to create looping songs.

If two Bxx commands are given, then the command in the higher channel will take effect.

### **Cxx** *Break to row.*

The Command Cxx signifies the end of the current pattern, and also that the next pattern should be played from row xx (hex).

If two Cxx commands are given... you know :)

### **D0x** *Volume slide down*

The volume slide down command causes the volume of the note to be reduced by x for each frame after (for that row). ie. If you have a note at volume 64, with command D01 and speed A06, then the final volume will be 64-5=59. A note at volume 32 with command D02 at speed A05 will result in a volume of 24.

For greater S3M compatibility, D0F will cause the volume to drop by 15 EVERY frame, instead of just off-note frames.

Here's a full frame-by-frame description which should provide greater understanding:

```
At 'speed' 4:  
C-4 01 .. D04
```

What this does is:

```
Frame 1: Plays C-4 with instrument 1  
Frame 2: Lowers the volume by 4  
Frame 3: Lowers the volume by 4  
Frame 4: Lowers the volume by 4
```

The next frame will be controlled by the next row of information.

### **Dx0** *Volume slide up.*

Operates exactly the same way as D0x, but slides the volume up by 'x'. Volumes cannot exceed 64 (checked and limited internally).

For greater S3M compatibility, DF0 will cause the volume to rise by 15 EVERY frame, instead of just off-note frames.

### **DFx** *Fine volume slide down.*

Takes the volume down by x at the start of the row.

**DxF** *Fine volume slide up.*

Takes the volume up by x at the start of the row.

**Exx** *Pitch slide down*

The pitch will slide down with speed xx (hex). In linear frequency mode, a pitch slide down by a particular value will always cause the same "pitch interval" - this does not occur in Amiga frequency mode. Valid ranges for xx are between 0 and 0DFh (as > 0E0h will be interpreted as fine slides)

**EFx** *Fine pitch slide down*

Slides the pitch down by x at the start of the row.

**EEx** *Extra fine pitch slide down*

Same as EFx, but 4 times finer. (ie. EE4 is equivalent to EF1)

**Fxx** *Pitch Slide up*

**FFx** *Fine Pitch slide up*

**FEx** *Extra fine pitch slide up*

Operates in the same manner as the Exx commands, but slides the pitch up. If the pitch gets 'too high', then the channel is turned off.

**Gxx** *Portamento to note.*

This command requires 2 parameters: A note to slide to and a speed. Example:

```
C-4 01 .. .00
G-4 01 .. G08
... .. G00
... .. G00
```

This will cause the note C-4 to slide to G-4 with speed 8.

You \*CAN\* change the sample that the note is sliding to. It is a good idea to make sure that the C5 Speed of the samples is similar, if you want to try this! The new sample will play from it's beginning.

**Hxy** *Vibrato with speed x, depth y*

Causes the frequency to oscillate with depth 'y' at speed 'x' which causes an interesting effect.

Don't get carried away with it though! If you specify EITHER x or y as 0, then the previous value will be used.

**Ixy** *Tremor with ontime x, offtime y.*

Causes the volume of the instrument to remain normal for x frames, then sets the volume to 0 for y frames.

**Jxy** *Arpeggio with halftones x, y.*

This causes the note to quickly cycle through three notes - the note playing, a note x halftones above, and a note y halftones above. This causes an effect similar to old C-64 chords.

Example:

```
C-4 01 .. J47 will cause the notes C-4, E-4 and
G-4 to be cycled.
```

**Kxx** *Dual command: Vibrato + Dxx (Volume slide)*

Note: The vibrato could have been set with either Hxx or Uxx

**Lxx** *Dual Command: G00 + Dxx (Portamento to and Volume slide)*

**Mxx** *Set channel volume*

Each channel has a volume range from 0->40h. The lower the value, the softer the notes in the channel. This command is convenient for making 'echoes'... you won't have to modify the echoing channel from the original, except for placing a M20 at the top of it!

Values greater than 40h are ignored at playtime.

**N0x, Nx0, Nfx, NxF** *Slide channel volume commands*

These commands work in the same manner as the slide volume commands, but operate on the channel volume, rather than directly on the note volume.

**Oxx** *Set sample offset.*

This will cause a sample to be played from offset yxx00h. This is useful to skip past the first part of a sample which may have a loud hit, or to start a speech sample half way through.

If you specify a value PAST the end of a sample, then the command is ignored.

The "y" part of yxx00h is set with command "SAy"

Example:

```
... .. SA5
C-4 01 .. 010
```

This will play note C-4, instrument 1 at offset 51000hex

Note to programmers: Oxx for 16-bit samples will move to the xx00h\*2 position in the sample - ie. the 'xx00th' sample.

**P0x, Px0, Pfx, PxF** *Slide panning commands.*

These work in the same maner as the slide volume commands, but operate on the channel panning. P0x slides the panning right, while Px0 slides the panning left.

**Qxy** *Retriggers a note after y frames with volume modifier x.*

This will cause a sample to be replayed from it's beginning after every y frames. 'x' can be any of the following:

<b>Value</b>	<b>Effect on volume each retrigger.</b>
0	No change
1	-1
2	-2
3	-4
4	-8
5	-16
6	*2/3
7	*1/2
8	No change
9	+1
A	+2
B	+4
C	+8
D	+16
E	*3/2
F	*2

**Rxy** *Tremelo with speed x, depth y*

This command acts similarly to the vibrato command, but affects the note volume, instead of the pitch. If you specify EITHER x or y as 0, then the previous value will be used.

- S3x** *Sets vibrato waveform to x*
- S4x** *Sets tremelo waveform to x*
- S5x** *Sets panbrello waveform to x*

Values for x:

- 0 = sine wave
- 1 = square wave
- 2 = Ramp down
- 3 = Random!

- S70** *Past note cut*
- S71** *Past note off*
- S72** *Past note fade*

With the loss of 1-1 correspondence of editing channels to internal playing channels, these commands provide control over notes that have already been played by a channel.

Example:

Consider the following situation, where instrument 1, notes C-4 & D-4 map to a looped sample, and the New Note Action is set to continue.

```
C-4 01 64 ...
D-4 01 64 ...
E-4 01 64 ... <--- Use S70 here!
```

Ordinarily, control would be lost over C-4 and D-4, but using command S70 at the point indicated, the notes C-4 and D-4 would be stopped immediately when the note E-4 plays.

Similarly, note off commands and Fadeout commands can also be issued.

Note: You **\*MUST\*** be in instrument mode for these to have any significance.

- S6x** *Pattern delay for x ticks.*

- S73** *Set NNA to note cut*
- S74** *Set NNA to continue*
- S75** *Set NNA to note off*
- S76** *Set NNA to note fade*

These options allow you to override the default NNA for a particular NOTE (ie. it does not affect the instrument) See section 2.5 for an explanation of NNAs.

Note: You **\*MUST\*** be in instrument mode for these to have any effect.

- S77** *Turn Volume envelope off.*

Stops the processing of the volume envelope. This is for the times that you don't want to use the volume envelope that you've created.

- S78** *Turn volume envelope on.*

Sometimes, you'll decide that it's better not to use the volume envelope for most cases, but you may just want to use it a few times - this is the command that lets you do that!

Note: You **\*MUST\*** be in instrument mode for this to work.

- S8x** *Set panning position*

Set panning position, S80 is leftmost, S8F is rightmost. This is an obsolete command, provided only for ST3 compatibility. Use command Xxx instead.

**S91** *Set surround sound!*

For those of you who are lucky enough to have a dolby surround sound decoder (myself not included - so if ya wanna send me something.....), this \*should\* cause the sound to come from the surround speaker. This is a pseudo-panning command, and issuing either Xxx or S8x will cause the surround sound to be reset.

If you try and play Surround Sound on a GUS, Interwave or AWE32, it will be played as a central pan.

**SAy** *Set high-offset. Check Oxx for an example.*

**SB0** *Set loopback point*

**SBx** *Loop x times.*

This pattern space-saving feature will cause the pattern to be looped x times back to the last SB0. Note that you can only loop within the pattern! Also, each channel has it's own loop-back information... so you HAVE to have the SB0 and SBx in the same channel for it to operate.

**SCx** *Note cut after x frames*

This will cause a note to be immediately stopped after x frames. It is similar to issuing a "^^^" in the note column, just that it has a finer control over timing.

**SDx** *Note delay for x frames*

Actually, this command works like "interpretation" delay for x frames. Any data - note, sample/instrument or volume in the channel will not be interpreted until x frames into the row.

**SEx** *Pattern delay for x rows*

This will cause a "pause" on the row for effectively x rows longer.

Note that if TWO pattern delay commands are issued, then the only the command in the higher channel will be considered.

**Txx** *Set tempo to xx*

Valid ranges are between 20h and 0FFh. The higher the value, the faster the playback. This essentially determines the time length of each frame, by the following formulas:

$$\text{Frames per minute} = 24 * \text{Tempo}$$

equivalently:

$$\text{Frames per second} = 0.4 * \text{Tempo}$$

**T0x** *Tempo slide down*

**T1x** *Tempo slide up*

Tempo slides up and down are used to smoothly modify the speed of the song. The tempo is modified by  $\pm x$  every non-row frame.

**Uxy** *Fine vibrato with speed x, depth y*

Same as vibrato, but 4 times finer in depth.

**Vxx** *Set Global volume*

Valid ranges are between 0 and 80h. All notes playing are affected by this change. Values greater than 80h are ignored at playtime.

**W0x, Wx0, WFx, WxF** *Slide global volume*

Similar to the Dxx commands, but operate on the global volume. Useful when fading out songs.

**Xxx** *Set panning position*

Sets the panning position anywhere from left to right (X00 is left, XFF is right).

**Yxy** *Panbrello with speed x, depth y.*

What the hell is panbrello? Well, it's a word that I coined late one night after a friend suggested that there should be a random pan position command. I decided.. why not - I'll make the vibrato/tremelo equivalent of panning! And so you have here, 'panbrello.' What this does is instead of oscillating frequency (pitch) or volume, it oscillates the panning position about the 'set' panning position for a channel. This is for you techno freaks out there who want another function to stuff around with! :)

The random pan position can be achieved by setting the waveform to 3 (ie. a S53 command). In this case *\*ONLY\**, the speed actually is interpreted as a delay in frames before another random value is found. so S14 will be a very QUICK panbrello, and S44 will be a slower panbrello. With any other waveform, the higher the value for x, the faster the panbrello, like vibrato and tremelo. If you don't know what I'm talking about, check out the stereo-indicators on the info page (don't forget that you have to be in stereo mode!)

If you specify EITHER x or y as 0, then the previous value will be used.

## 5) Editing Functions

For a complete list of available functions, check the help screen for the pattern editor (on **F1**)

<b>Ctrl-Backspace</b>	IMPORTANT! This one will save you a lot of frustration - it's a 10-stage listed Undo function!
<b>Grey Plus</b>	Advance to next pattern
<b>Grey Minus</b>	Goto previous pattern
<b>Shift-GPlus</b>	Advance 4 patterns
<b>Shift-GMinus</b>	Go back 4 patterns
<b>Ctrl-GPlus</b>	Go to the next order's pattern
<b>Ctrl-GMinus</b>	Go to the previous order's pattern
<b>Alt-Delete</b>	Remove an entire row from the pattern
<b>Alt-Insert</b>	Insert an entire row into the pattern
<b>'.'</b>	erase data.
<b>Space</b>	will enter the previous data for that column.
<b>'4'</b>	play the note under the cursor.
<b>'8'</b>	play entire row.
<b>Ctrl-F6</b>	play pattern from the current row.
<b>Ctrl-F7</b>	Set playback mark - this will be the position that subsequent playback will occur when you press <b>F7</b> . Remove the mark by pressing <b>Ctrl-F7</b> on the row that is already 'marked.'
<b>Alt-0 -&gt; 9</b>	Quick select "skip value" The skip value is the number of rows that the editor advances by when a note is entered - this makes it easy to enter notes on alternate rows (press <b>Alt-2</b> first!)

If the skip value is 0, then the editor will advance to the next channel making it easier to enter chords)

If the skip value is 0, AND template mode is on, then entering a note will cause the editor to advance by the length of the template block.

<b>Alt-Enter</b>	Store current pattern in memory
<b>Alt-Backspace</b>	Restore current pattern

## 6) Block Functions

<b>Alt-B</b>	Mark top left of block
--------------	------------------------

<b>Alt-E</b>	Mark bottom right of block
<b>Alt-L</b>	Mark entire column (channel)
<b>Alt-L*2</b>	Mark entire pattern
<b>Alt-D</b>	Mark the minor row highlight number of rows. Pressing this repeatedly doubles the length of the block. eg. Pressing <b>Alt-D</b> once may mark out 16 rows, pressing it twice will mark 16 rows.
<b>Alt-A</b>	Transpose all the notes in the block a semitone down If no block is marked, then the current note will be transposed a semitone down.
<b>Alt-C</b>	Copy marked block into the clipboard
<b>Alt-F</b>	Double the length of the selected block
<b>Alt-G</b>	Halve the length of the selected block
<b>Alt-I</b>	Toggle Template mode... read part 8
<b>Alt-J</b>	Volume Amplifier... modify the volume controls between 0 and 200%
<b>Alt-K</b>	Slide volume... if you want to manually control a fadeout, or fadein, then set the volumes at the extreme ends of the blocks, then press <b>Alt-K</b> ONCE!
<b>Alt-K*2</b>	Remove all volume controls in the selected block
<b>Alt-M</b>	Mix clipboard with pattern data.. try it out, and you'll understand (it's like an advanced paste command - it'll only copy the data IF there is nothing on the row it is copying to.) Pressing this two times will copy the data if there's nothing in the same FIELD (not just same row).
<b>Alt-N</b>	Toggle Multichannel mode.. Multichannel mode allows you to select for each channel whether it should be part of the multichannel list. If you enter a note while on a channel which is in the multichannel list, then the editor will advance to the next channel.
<b>Alt-N*2</b>	Multichannel menu.
<b>Alt-O</b>	Overwrite pattern data with clipboard. Like paste, but without the "insert" part :)
<b>Alt-P</b>	Paste clipboard into pattern at current position
<b>Alt-Q</b>	Transpose all the notes in the block a semitone up If no block is marked, then the current note will be transposed a semitone up.
<b>Alt-S</b>	Set all the instrument (sample) values to the current instrument (sample) within a block
<b>Alt-U</b>	Unmark block / release clipboard from memory.
<b>Alt-V</b>	Set all the volume controls to the current
<b>Alt-W</b>	Remove all volumes not associated with a note/inst
<b>Alt-X</b>	Slide command value. Similar to <b>Alt-K</b> (slide volume). This command can be convenient to slide the panning from one position to another, or to slide global volume effects, channel volume, sample offsets... whatever...
<b>Alt-X*2</b>	Erase all effect data in the selected block!
<b>Alt-Y</b>	Swap selected block with a samesized/shaped block starting at the current position. Swap blocks CANNOT overlap.
<b>Alt-Z</b>	Cut current block. Warning: If you don't have enough memory, the current block WILL be erased, although there is no record of it in the clipboard....

## 7) Track View functions

Sometimes you just want to know what you've put in those tracks that are currently off the screen... these functions allow you to do just that!

<b>Alt-T</b>	Cycle Track view mode.
<b>Alt-R</b>	Remove all track views
<b>Alt-H</b>	Remove track view divisions (allows you to view 36 channels!)
<b>Ctrl-0</b>	Remove current track from track view
<b>Ctrl-1-&gt;5</b>	Quick select a track view for current channel.

## 8) Templates



Templates are an invention to make entering repeated 'sequences' easier. Often, you may want to enter a note with the same effects following each note. eg.

```

C-4 01 00 D10  }
... .. D00  }
... .. D00  } Enter this....
... .. D00  }
... .. D02  }
... .. D00  }
D-4 01 00 D10  }
... .. D00  }
... .. D00  }
... .. D00  }
... .. D02  }
... .. D00  } Use the templates to enter this!
E-4 01 00 D10  }
... .. D00  }
... .. D00  }
... .. D00  }
... .. D02  }
... .. D00  }
etc..

```

With templates, this can be done by entering the first 'block', then marking it and copying it into the clipboard (using **Alt-C**). Toggle the template mode by pressing **Alt-I** to Template: Overwrite, then go to the start of the second block, and press the key for D-4 (normally X). The whole block will be filled in! This is easiest to understand if you print this section out and try it in the editor.

The templates can also include other notes, which will be translated accordingly, and can span several channels.

A VERY neat use of templates is to enter multiple notes: (eg. octaves, different samples, left/right stereo sample pairs...)

```

C-4 01 32 .00   C-5 02 32 .00  } Enter this...
G-4 01 32 .00   G-5 02 32 .00   }
D#4 01 32 .00   D#5 02 32 .00   }
C-4 01 32 .00   C-5 02 32 .00   } And use Templates here!
G-4 01 32 .00   G-5 02 32 .00   }
D#4 01 32 .00   D#5 02 32 .00   }

```

Templates can span more than one row and one channel - you can template blocks of any size.

### 2.3 Order List, Channel panning & volume. (F11)

After creating patterns, the tracker has to know what order to play them back in. The order list is the means by which this is done. Another wording of the **Order List** which may make it simpler to understand is the **Pattern sequence list**. ie. the sequence in which the patterns are played back. Simply enter the patterns into the list in the order they should be played, and voila! - you have a new piece of music. (as long as you have something in the patterns, of course!). The "---" marks the end of a song, and "+++" is simply a marker which is skipped (for ST3 support). Pressing **N** on a row will enter the previous order's pattern+1. eg. Entering 000, then pressing **N** 5 times will enter 000 (which you typed), then 1, 2, 3, 4 and 5.

Initial channel panning and volume is the panning and volume that each channel is set to whenever the song is reset. It will not affect anything currently playing. On the panning list, you can also mute channels by pressing spacebar. Pressing **S** will select the initial 'panning' as surround sound. The initial channel volumes can be accessed by pressing **F11** once you are already on the **Order list** and channel panning screen.

## 2.4 Samples

### 2.4.1 Information about Samples

Samples are the raw sound information. They can be of *\*anything\** you like - normally a musical instrument of some sort, but you could have voice samples or sound effects too!

*Impulse Tracker* accepts 8-bit or 16-bit samples. It can import several different formats, which are detailed in section 1.4

If you import a 16-bit file which isn't recognised, you have to use the following steps:

- 1) Press **Alt-A** (to convert signed/unsigned) and convert the data.
- 2) Press **Alt-Q** (to toggle to 16-bit) and DON'T convert the data
- 3) If the sample is unsigned, you will then need to convert it AGAIN by pressing **Alt-A** (and selecting yes to convert data)

Samples have the following information: A name, a default volume, a global volume, vibrato information, loop information and a frequency

The default volume of a sample is the volume at which a sample is played, IF no volume is specifier.

The global volume is the modifier by which ALL occurrences of the current sample are scaled by.

Vibrato information - the vibrato speed is the speed at which the vibrato oscillates. Although it is possible to enter values 0->64, the best effects are obtained by entering values around 24-48. The vibrato depth determines the maximum deviation from the mean frequency. The vibrato rate determines how quickly the vibrato is applied. Higher values cause the vibrato to be applied more quickly, a low value like 1 can take many seconds before the vibrato is noticed.

Loop information. Samples can be looped to provide a continuous sound. The loop beginning and the loop end must be specified, and the loop method (either forwards or ping pong) must be set. (Toggle by pressing spacebar). The loop boundaries can also be modified using '+' or '-' (even while the sample is playing!)

Sustain loops are the loops used as long as no note off command is encountered. In the sample tester, releasing a key provides a note off command to the player. Once a note off command is issued, the normal loops apply. The idea behind sustain loops originated after listening to some synthesizers which provided 'after note' sounds - like fingers lifting off guitar strings. With sustain loops, you can have a guitar sound in the sustain loop, with the end of the sample containing the 'finger off' sound. This way, you should be able to achieve a new degree of realism. The best way to do this is to have a reasonably small sustain loop, and a section after it containing the note-off sound, possibly looped. (The sustain loop should ideally be less than 1/20th of a second, so that you have reasonably good control over the timing of the note-off, but this is not always possible)

Example: If you have a sample with no sample loop, but a sustain loop around bytes 19000-20000, then while no noteoff command is issued, then the sample will loop between 19000 and 20000. Once a noteoff command is received, then the sample will be played until it's end and then will stop.

If you have a sample of length 64000 with a ping pong loop around the whole sample (0->64000) and a sustain ping pong loop around bytes 20000-30000, then the sample will loop backwards and forwards between 20000 and 30000 until a noteoff command is received, after which it will loop between 0 and 64000 (backwards and forwards).

It is recommended that you do NOT have a ping pong sustain loop going to a forwards loop or no loop and that if you DO have a sustain loop AND a normal loop, that the sustain loop lies within the normal loop (for ping pong), or that the normal loop end is after the sustain loop end (for forward's loop) (think about what you're asking it to do in these cases carefully... it should make sense - the tracker can still handle the cases when this is not adhered to, but the resulting note-off effect can vary greatly depending on timing and pitch!)

The frequency of a sample determines its pitch. The higher the frequency, the higher the pitch. The frequency can be doubled or halved by pressing **Alt-Grey Plus/Minus** to cause an octave rise/decrease. For the technically inclined out there, the frequency is the number of bytes per second that have to be played for a C-5. You can increase the frequency by one semitone by pressing **Ctrl-Grey Plus** or decrease it by a semitone with **Ctrl-Grey Minus**.

To load in a sample, go to the entry in which you want to load it then press enter. The sample library should appear. You can test out any sample just by "playing" notes on the keyboard while you are on top of the sample. Load a sample into the sample list by pressing enter. Of course, if you don't have any samples, no samples will appear :)

Where can I get samples from??

You can steal ('rip') samples from modules by going to the sample in the sample list, then pressing **Alt-O**. Scream Tracker 3 samples can be saved by pressing **Alt-T** (vibrato, global volume + susloop information lost) or raw samples can be saved by pressing **Alt-W** (all variables lost!)

Warning: Be careful where you rip samples from!! Some people don't welcome it at all! And in some cases, sample ripping may be a breach of copyright - this mainly a concern when you use commercial samples)

There are also many great sample 'packs' available... try a local music BBS, or the bigger FTP sites.

Some synthesizers have patch files (eg. Kurzweil 2000 as .KRZ) which are available through the internet. You can download these, and convert them with an appropriate utility (I recommend Convert 1.4 - convrt14.zip on ftp.cdrom.com/pub/demos/programs/convert and use it to convert the files to .S3I format - even for 16 bit samples). I don't know what sort of copyright these samples have. \*YOU\* have to read any text accompanying the files to find this out.

Finally, you can make your OWN samples! With a sampler (eg. *Digiplay 3.0*) or with some other tracker (eg. *Fast Tracker 2*), you are able to record information through the microphone socket of the computer!

Note: The length of a sample is \*NOT\* necessarily the number of bytes it occupies! The length quoted is the quantity of SAMPLE INFORMATION. If you have an 8 bit sample, then the size in bytes IS the same as the length. If you have a 16-bit sample, then the size in bytes is DOUBLE the length.

## 2.4.2 Sample Functions

**Alt-A** and **Alt-Q** provide basic sample-conversion functions. **Alt-A** will convert a sample to or from unsigned to signed format. **Alt-Q** will allow you to change a sample between 8 and 16 bits.

If you want to remove the part of a sample before a loop or after the end of a loop, you can use Pre-loop cut sample (**Alt-B**) or Post-Loop cut sample (**Alt-L**)

To reverse a sample, use **Alt-G**. This can produce interesting effects.

If you are working to a size limit, check out the functions **Alt-E** and **Alt-F**. These functions allow you to resize the sample to whatever size you would like! (Note that there is a reduction in quality associated with a reduction in size). **Alt-E** will resize the samples WITH interpolation, **Alt-F** will resize the samples WITHOUT interpolation.

To decrease the volume of a sample, the best way for 8 bit samples is to reduce the Sample's GLOBAL volume. For 16-bit samples, it's better to just attenuate the sample (ie. use **Alt-M** to 75%). The sample will retain greater quality if you follow these guidelines.

**Alt-M** is used to amplify a sample to between 0 and 400%. The default value for **Alt-M** which appears is the maximum the sample can be modified without creating clipping distortions.

To exchange two samples \*in the sample list only\*, use **Alt-X**. To swap two samples \*even in the pattern data\*, use **Alt-S**.

## 2.5 Instruments

For those who have never used a tracker before, I strongly suggest that you skip this section for now - learn how to 'use' samples first. After that, you'll should be able to understand and hopefully fully appreciate the power that instruments can provide.

To enable instruments, go to the song variables (**F12**), and press on the "Instruments" button after then "Control" prompt. This \*MUST\* be done in order to use ANY instrument function (including special note effects, viz S7x).

The instrument parameters are split into four screens - one for each of general options, volume options, panning options and pitch options. To select the appropriate screen, just whack enter on any of the four buttons at the top of the instrument list.

Instruments are collections of samples (or just a single sample). The translation of samples is controlled by the column in the centre of the screen on the general options page which shows what note/sample pair a single instrument note will be translated to.

Example: If you're on instrument 1, and the note translation table appears as such:

```
C-5 | C-5 01
C#5 | C-5 02
D-5 | D-4 03
D#5 | D#4 03
```

Then entering "C-5 01" into a pattern will cause sample 1 to be played at pitch C-5, "C#5 01", will cause sample 2, to be played a pitch C-5, "D-5 01" will cause Sample 3 to play at D-4, "D#5 01" will cause Sample 3 pitch D#4 to play... get the idea?

Why would you want to combine several samples into one instrument? Well, one very good reason is that it makes it easier to enter drum parts - you no longer have to change the 'instrument' if you want another sample. And also, combining them into one instrument makes them easier to manage - you can specify a volume envelope, NNA or FadeOut which will apply for ALL of the samples in the list.

Another reason is if you want a very high quality sounding 'instrument.' If you wanted to go overboard, you could sample every single note on a piano separately, and set up the note translation table to point to each sample accordingly - you'd get a very, very nice sound - if you don't run out of memory first. Or patience :)

When you save an instrument to disk, all the related samples are stored with it. This means that you can setup a 'drum kit' instrument, and to load it into another song, you only need to select that instrument and all the samples will be loaded for you.

The Fadeout value for each instrument determines how quickly the volume of the instrument decays under any of the following conditions:

- 1) NNA "FadeOut" is selected, and another note is played. (see later in this section for info about NNAs)
- 2) The end of a volume envelope is reached.
- 3) A note off command is encountered, without a volume envelope.
- 4) A note off command is encountered, and the 'normal' volume envelope loop is on.

The larger the fadeout value, the quicker the volume decays.

Each instrument also has an associated volume/pan/pitch envelope! The volume envelope is edited by selecting the node (left/right arrows) then picking it up (with enter), moving it around (using arrows, or **Alt-Arrows** for quicker control), then pressing enter again to "put down the node". Nodes can be inserted or

deleted (with the keys insert and delete!). Note that the envelope will only be used if the envelope flag is set to "on" (just below the envelope graph)..

Remember that you have to be in instrument mode for this all to work!

There are 3 pieces of information for each Envelope graph - the node number (hopefully obvious), the time of the node (the number of 'ticks' or 'frames' that elapse before the node point) and the value of a node (also hopefully obvious). So the x-axis is time, and the y-axis is volume/pan/pitch - simple, really!

You can also specify envelope loops and sustain loops. These operate in the same manner as the sample loops, but the numbers refer to node numbers.

The powerful feature of this tracker, though, is not the envelopes by themselves - it's the New Note Actions! What these options do is allow you to select what should happen to an instrument when another note is played in the same column. If NNA "Cut" is selected, then the previous note will immediately be stopped (like in all other trackers). If NNA "Continue" is selected, then the note will continue playing! This is especially useful for Drum Parts, where there may be a long-ish snare sample - you can go on ahead and put a bass drum in the very next row after it - the snare drum will still complete playing! NNA "Note Off" issues a note off command to a note when a new note is played in the channel. This is particularly useful in combination with volume envelopes and volume envelope sustain loops. NNA "Note Fade" causes the current note to fade out with the fadeout value when a new note is played.

### **WARNING!!**

New Note Actions are EXTREMELY powerful, but they CAN cause problems if you are not careful. Selecting NNA Fade with a fade value of 0, or note continue with a looped sample (no volume envelope) or anything which can cause a quick build up of allocated channels can easily hang a slow computer (or even a quick one, for that matter!) I tried to put checks against this, but they always triggered too late - "past the point of no return" where the CPU becomes so bogged down with processing the information, that it can't do or try to do anything else. (The tracker has \*MANY\* calculations to do - the most time consuming are the mixing routines, which require the processor to process as many bytes as the mixing speed per second for EACH note ie. a mixing speed of 44kHz means that for each note playing, 44000 calculations have to be made EVERY second... so with 64 channels 'active' at the maximum mixing rate for a *SBl6*, almost 3 MILLION calculations have to be done EVERY second to produce the sound (in mono)!!! (My 486 can cope with this, but my 386 just dies!)

Duplicate Check Type (DCT) / Duplicate Check Action (DCA)

DCT = Off/Note/Sample/Instrument, DCA = Cut/Off/Fade

When the duplicate check type is enabled, then repetitions of the same instrument&note/sample (or just repetitions of the same instrument) pair in a particular channel will cause the previous occurrence of the instrument&note/sample pair to be cut or faded (depending on the DCA)

Example: If the DCT is set to note, and DCA is set to Cut, then the asterixed notes will cut out the tilda notes

```
+-----+
| C-4 01 ~ | | C-4 02 ~ | | C-4 03 } |
| D-4 01 | | C-4 02 * ~ | | D-4 03 } Nothing get's cut. |
| C-4 01 * | | C-4 02 * | | C-4 02 } |
+-----+
```

This option was included to help limit the number of active channels, and is especially useful for drum tracks.

Another nifty application of Duplicity checks is the following example: You can have a separate sample for each string of a guitar and setup and instrument to accomodate this. Setting the NNA to continue, the DCT to sample and the DCA to fade (with a relatively quick fadeout) means that whenever you play a new note, the previous note will continue to play on. BUT! If a previous note of the same instrument has the same sample as

the new note being played, then it will be faded out - this closely relates to what you hear when someone plays a guitar - when they play the same string, they have to put their fingers down on the string, which causes the last note \*on that string\* to fade out.

I sincerely hope that these options do not cause the death of brilliant 4 channel music - that would be a great shame! There is truly an art in making a decent sound in as few channels as possible! (If you're new to this sorta thing, then it'll grow upon you...)

For interest's sake...

Channels are turned off internally under any of the following conditions:

- 1) The end of a sample is reached (quite obvious)
- 2) When the end of a volume envelope is reached, and the final envelope volume is 0
- 3) When the fadeout value for a channel causes it to become silent.
- 4) When a duplicate note is played when DNT is set to Note and DCA is set to cut for the instrument.
- 5) When a notecut is issued (obvious)
- 6) When a channel is moved to the background (using NNAs) AND the volume is 0.

Out of all the functions provided for the Samples and Instruments, perhaps the only one that requires explanation is the "Update Pattern Data" function. This function was written for people who have already written music in MOD/S3M/MTM formats, and want to combine their percussion parts into a single instrument. The way to do this is to set up the instrument->note/sample table, and then use the "Update Pattern Data" function. What this does is search through all the patterns for all occurrences of the note/sample pairs that appear in the instrument->note/sample table, and replace it with the appropriate note/INSTRUMENT pair.

Sounds quite complicated... sorry :)

Example:

If you originally have the following setup

Sample 1 = Bass Drum  
Sample 2 = Snare Drum  
Sample 3 = Closed Hihat  
Sample 4 = Open Hihat

And the following column within any pattern:

```
C-5 01 .. ...  
C-5 03 .. ...  
C-5 03 .. ...  
C-5 03 .. ...  
C-5 02 .. ...  
C-5 04 .. ...  
C-5 01 .. ...  
C-5 04 .. ...  
C-5 01 .. ...
```

You can combine these four samples into one instrument by creating the instrument->note/sample table with the following entries: (say for instrument 10)

```
C-5;C-5 01  
C#5;C-5 .. <--- just an empty slot...  
D-5;C-5 02  
D#5;C-5 ..  
E-5;C-5 03  
F-5;C-5 04
```

And using the update pattern data command will produce the following:

```
C-5 10 .. ...
```

```

E-5 10 .. ...
E-5 10 .. ...
E-5 10 .. ...
D-5 10 .. ...
F-5 10 .. ...
C-5 10 .. ...
F-5 10 .. ...
C-5 10 .. ...

```

## 2.6 Song Variables

Most of the options on this screen should be quite obvious, but they are explained here for clarity.

*Song Name* - should be obvious. Shove whatever you like in here :)

*Initial tempo* - The tempo that the piece starts with. The tempo calculation is explained under the set tempo command (Txx)

*Initial speed* - The speed that the song starts with. The speed calculation is explained under the set speed command (Axx)

*Global volume* - This scales all the volumes in the song. It may be necessary to change this value if there is overload occurring (in the form of crackles in the music) - especially on the Gravis UltraSound, as it is a non-mixing device, and not scaled by the mixing volume.

*Mixing volume* - This value affects mixing sound devices (ie. all Sound Blaster cards + PC Speaker).

*Separation* - The separation determines how far apart the left/right panning sounds. It is suggested that for a stereo system (where the speakers are relatively close together) that the separation be around its maximum (ie. 128). For a stereo system where the speakers are placed a large distance apart, a value of 80 may suit better. A suggested separation for headphones is around 40. This value has no significance unless stereo playback is selected.

*Old Effects* - When Impulse Tracker was first written, some effects were interpreted differently from other formats, most notably vibrato. When you turn this ON, then its effects will be interpreted how they used to be in ST3/MMEdit/... but when it's off, it'll operate how it used to in previous versions.

Differences:

The Vibrato (and Tremelo) used in IT is smoother than how it was implemented in MOD/S3M/etc. It is updated EVERY frame and hence is independent of song speed, whereas the standard vibrato WAS dependant upon song speed. Vibrato is two times 'deeper' with Old Effects on.

Sample offset commands past the end of a sample in IT were ignored, whereas with Old Effects on, the sample is played from its end point.

*Control* - This is the option that lets you use Instruments!

When Control Sample is selected, then all the instrument information is ignored. The query to initialise instruments, if accepted, will copy all the sample names to the instruments (if the samples exist) and set up the Note Translation Table for each instrument to point to a sample.

*Playback - Mono*: When you select mono playback, all stereo commands are essentially ignored. If you are using a *SBPro*, selecting Mono provides a higher quality output than selecting Stereo and using a central pan.

Stereo: This is the option to go for!

Note that stereo mode requires more processing power than mono (unless you are using a hardware driver)

*Pitch Slides* - Amiga: This is the mode to choose for compatibility with S3Ms, MODs, MTMs, XMs... when you select amiga pitch slides, you'll often have to experiment to find the adequate slide value.

- Linear: With the linear slides, a certain slide value will always cause the same music "interval" change. An example of what this means is if you slide up a note with speed 8 (ie. F08) at speed 5, then the result will be that the note is raised a tone. Everytime. So a C-4 will rise to a D-4, C-5 will rise to a D-5. (It's not like this with amiga slides! A C-4 \*MAY\* rise to a D-4, but if it does then the C-5 will rise to a note around E-5!!)

The benefit of linear slides is that if you slide a pair of notes at the same speed, they will maintain the same interval (pitch difference) throughout. Also, if you modulate (transpose) a pattern, you will not have to modify any slide values.

*Directories* - this shows the current song/sample/instrument directories. If you want to save the current directories as the default to have on bootup, then press enter on the Save button. Note that this save function also saves the palette, keyboard type, info page layout and pattern editing preferences to IT.CFG

## 2.7 Screem Tracker 3 Users

This section is just for users of Screem Tracker 3 who would like to know the differences between ST3 and IT without working through the whole document. In a nutshell:

1) Differences in use.

a) The **Order List**, Panning and Variables used to be on **F1**. In IT, **F1** has been reserved for the **Help Screen**, **F11** for the order list and panning, **F12** for the variables. If you REALLY do find this too inconvenient, use the command line switch -k to swap the interpretation of the keys **F1** and **F11**... no text will be updated tho (and it does seem sorta weird!)

b) Samples now have associated Vibrato information and a global volume. The global volume affects all instances of the sample throughout the song. This is useful when you replace a sample, which is at a different volume from the original - you can just adjust the global volumes of the samples to suit.

d) Block functions are no longer restricted to one column. This also means that when you copy entire patterns, you will have to go to the top left of the pattern. Also, you will have to set the number of rows in the destination pattern to the appropriate number if it is different from the source. This difference may take some getting used to - you may often find yourself doing several block functions in a single columns when you could actually have done it across the whole block! \* To mark blocks, you can use **Alt-B** and **Alt-E**, or **Shift+Movement** \*

e) In the pattern editor, the following commands have changed:

1) **Alt-T** : **Track View** has been upgraded. **Alt-T** will cycle through all of the possible 5 view methods.

You can use **Ctrl-1** to **Ctrl-5** to quick select a view, or **Ctrl-0** to remove a view. Press **Left-Ctrl+Shift 1->4** to select a 'different way to edit' :)

2) **Alt-X** : Pressing **Alt-X** once will slide the effect data value in a block (useful for controlling global volume, panning or sample offsets in some cases). Pressing **Alt-X** twice will cause all of the effect data to be deleted as in *Screem Tracker 3*

3) **Alt-K** : Pressing **Alt-K** once will slide the volumes. Pressing **Alt-K** TWICE will remove all volume controls in the block. (Works like **Alt-X**, but on the volume column)

4) **Alt-I** : Cycle Template control.

5) **Alt-J** : Volume amplifier. Select an amplification between 0 and 200%.

6) **Alt-H** : Now toggles view divisions on/off. Makes it possible to view/edit up to 36 channels in the pattern editor!!!

7) **Alt-Z** : Originally the Zap command (if pressed twice)... now it is effectively a "Block Cut" command. It will copy the block into the clipboard, and then wipe the block clean.

8) **Alt-V** : Will set all the volumes in the block to the default.

9) **Alt-W** : Will remove ALL volumes not attached to a note/instrument Therefore, the old **Alt-V** command can be simulated using **Alt-V** then **Alt-W**. The reason for doing this was so that the **Alt-K**, **Alt-W** combination would slide all the volumes with notes/instruments associated.



f) There is no chord edit feature in IT. I considered the chord edit feature of ST3 useless and a bad influence on new trackers :)

g) Patterns can be from 32 to 200 rows! (Under Pattern edit config on **F2**, or **Ctrl-F2**)

h) Pressing spacebar in the pattern editor will cause the previously entered note/instrument/volume/effect/effectvalue to be entered. Quite convenient. Hopefully.

i) **Alt-F10** now solos a channel instead of toggling all of the channels on/off. On the **InfoPage**, you can use 'S' to solo channels, and 'Q' to toggle channels. I felt that this was more usable than the original.

j) The infopage itself has become upgraded... press **PgUp/PgDn** to cycle through view methods, **Insert** to add a window, **Delete** to remove the currently highlighted window, **Tab** to move between windows, **Alt-Up/Down** to move the base of the window up/down.

k) To increase the sample frequency by an octave in the sample list, press **Alt-Grey '+'** or **Alt-Grey '-'** to decrease the frequency by an octave. To change it by semi-tones, use **Ctrl-Grey '+'** and **Ctrl-Grey '-'**

l) The Sample library is accessible from all screens in the program by pressing **Ctrl-F3**. The Instrument library is accessible on **Ctrl-F4**.

m) Each channel has an associated volume. Echoes can be created by just copying one channel onto another, then setting the "channel volume" (Command Mxx)

n) When channels are muted, all commands are still interpreted, but the notes aren't played. What this means is that commands such as Axx, Bxx, Cxx, SBx, Vxx, Txx, Mxx, Nxx will be processed even if they are in a muted channel.

o) Pressing **Ctrl-Grey Plus** and **Ctrl-Grey Minus** in the pattern editor will go to the next/previous pattern according to the order list!

p) Samples have sustain loops + Ping Pong loops supported

q) Note off command & Note cut command implemented... Note cut is exactly like ST3's ^^^ (and appears like it too!) Note off appears as "---" and releases sustain points.

r) You can enter panning values (between 0 and 64) in the volume column. Do this by pressing "'" to toggle between entry of panning/volume values.

s) You can use samples >64k and 16 bit samples!!!

t) When you use a skip value of 0, the cursor will move across to the next channel. This is a very convenient way to enter chords, especially when combined with the multi-channel selection (whack **Alt-N** twice)

## 2) Differences in interpretation of notedata.

a) Global volume changes will affect ALL of the notes playing. In ST3, global volume changes only affected new notes. The range for the global volume command is from 0 to 80h

b) If you specify an instrument without a note, then in ST3, the volume is set to the default volume. In IT, the volume will be set to the default volume \*IF\* the instrument specified is the SAME as the currently playing instrument. Otherwise, the new sample will be played from it's beginning.

c) Command X (Set panning, not REALLY implemented in ST3) now has range from 0 to 0FFh.

d) Command C (Break to row) now works in HEX. This is so that you can jump to any row in the (next) pattern.

e) New commands:

M:	Set channel volume
N:	Slide channel volume
P:	Slide panning
S5x:	Set panbrello waveform and reset panbrello position
S6x:	Pattern delay for x frames.
S7x:	Instrument related controls (8 different controls here)
S91:	Surround sound on!
SAy:	Set high offset
T0x:	Tempo slide down
T1x:	Tempo slide up.
W:	Global volume slide.
X:	Set pan position.
Yxy:	Panbrello

f) If you use the portamento to command (Gxx) to a different sample in ST3, the sample offset is undetermined. In Impulse Tracker, the sample will be played from it's start.

g) The commands Hxx and Uxx will cause a vibrato half the depth of ST3 if "Old Effects" option is OFF and the vibrato in IT is also smoother than the vibrato in ST3.

3) The greatest difference is the implementation of instruments. Read section 2.5 for more information.

4) Saving S3M modules.

The following guidelines have to be followed to compose 100% compatible S3M modules.

#### General Stuff

- 1) \*Pitch slide mode MUST be amiga.
- 2) \*All initial channel volumes MUST be 64
- 3) \*No instruments functions can be used. If you ARE in instrument mode, however, Impulse Tracker will translate the pattern data according to the Note Translation Tables.
- 4) \*The song message is \*NOT\* saved in S3M format.

#### Pattern stuff

- 1) \*The total number of patterns MUST NOT exceed 100.
- 2) \*The number of rows in each pattern MUST be 64
- 3) \*Data is only translated within the first 16 channels.
- 4) \*All notes must lie within the range of C-1 to B-8
- 5) \*Several commands will not be interpreted by a standard S3M player (viz. Mxx, Nxx, Pxx, S5x, S6x, S7x, S91, SAy, Wxx, Yxx). Also, you should have Old Effects (on the variable list) put to \*ON\* if you want to write S3M files.
- 6) \*Panning controls in the volume column are NOT stored.

#### Sample stuff

- 1) \*No sustain or ping pong loops can be used
- 2) \*No sample vibrato can be used
- 3) \*The sample global volume must be 64.
- 4) The Scream Tracker 3 module format supports 16 bit samples and samples >64k although the actual program did NOT. Note that \*MOST\* players probably ignore these fields and it is likely that the module will NOT be played correctly by a standard S3M player if it contains samples >64k and/or 16 bit samples.

All conditions with an asterix preceding them are checked at save time.

Note that both note off (---) and note cut (^^^ ) will get translated to a note cut (^^^ ) in S3M format.

## 2.8 Gravis UltraSound / Interwave / AWE32 users.

For hardware mixed devices (eg. *Gravis UltraSound*, *Interwave* and *AWE32*), there are some limitations that are outlined here.

I've already mentioned a few things throughout the DOC about hardware mixing, but I'll reiterate them here with a few more points.

- 1) The maximum number of notes that you CAN play with a *GUS* and *Interwave* is 32. The maximum number of the *AWE32* is \*30\*.

This was incorporated by effectively limiting the number of active channels to 32/30. So if you play a 64 channel song which is using control:Samples, then you will LOSE the last 32/34 channels. If you are playing a 64 channel song with control:Instruments, you shouldn't miss out on too much. This shouldn't really be a problem in 99.9% of the cases.

- 2) The *GUS* playback quality is dependent on the maximum limit of active channels. The *GUS* can be initialised to play anywhere between 14 and 32 channels. Values above 32 are reduced to 32, and values below 14 are increased to 14. The lower the number of channels, the higher the playback quality. (14

channels will give 44100 Hz, with 16-bit interpolation - very, very nice quality!). The default *GUS* driver dynamically chooses a particular number of channels. This doesn't ALWAYS work with all cards, so if you don't like the transition of sample quality as the channels increase, or your GUS seems to stop notes unexpectedly, use the other driver(s).

3) The *GUS/Interwave/AWE32* routines occupy less memory than the mixing routines. You will have about 30k-100k more of memory to play around with.

4) The memory allocation routines are very simple and not very versatile. When you are working through the sample library, samples are NOT always deallocated when you move on to the next sample! If you get the message "Out of soundcard RAM", then it may be necessary to press **Ctrl-G** (or select Reload Samples from the samples menu) to reorganise the samples in memory. Sorry about the inconvenience.

5) Changing the mixing volume does NOT affect these cards. It may be necessary to reduce the global volume to remove crackle if the output is overloading.

6) The surround sound option will cause the sample to be played with a central panning. This is because I don't know how to get hardware surround without wasting a lot of memory and channels.

7) The panning positions are reduced to a scale of 0->15 in the GUS, so fine alterations in the panning may go unnoticed. The *Interwave* and *AWE32* permit 256 pan positions

### **3. Before you write to me. (Important notes)**

I would very much like to get feedback on this program, but I am already aware of several problems (?).

1) If you're going to write because the program looks 'too' much like *Scream Tracker 3*, then don't :) I've used *Scream Tracker*. I loved using it - it was the best, easiest, most convenient program ever, so why not adopt it's design????

2) The program could easily crash if you try to load a corrupted module. If you've got some modules that you know are corrupted, try to resist the temptation of testing the stability of the program.... in many cases it'll die :)

3) A couple of people have asked about ASCII characters > 128. I'm sorry - I can't let you have 'em on anywhere else except the message editor. The reason is because I've used a lot of the characters for 'graphics' - real time character generation. (the ASCII characters > 128 don't really exist anymore)

4) If you 'overload' your CPU... the subsequent playback can also become distorted! To fix this, reinitialise your sound driver with **Ctrl-I**.

5) The help screens have only been defined for the following screens:

- a) Pattern Editor
- b) Sample List
- c) Instrument List
- d) Info Page
- e) Order list and panning
- f) Order list and channel volumes
- g) Message editor

I didn't think it necessary to include help screens for the other screens, but if you feel strongly about including help on certain points/screens, then I suppose I could make the effort...

6) If you have a bug to report, please check through BUGS.TXT first to see if it's listed, and also check through this document - the problem may be in the understanding of how the program works! Otherwise, contact me, AND LET ME KNOW WHAT VERSION YOU ARE USING!

7) Any further *GUS* click removal is very difficult to achieve. ST3's *GUS* click removal requires double the number of channels on the *GUS* to be used and I can't allocate any more! Try out the alternative *GUS* drivers first

8) Got suggestions for IT? At the moment, I'm really sorry - I barely have enough spare time myself. I can't promise to add any suggestions you make, but please send them along anyway.

You can contact me by writing to:

Jeffrey Lim  
9 Wilgena Avenue  
Myrtle Bank  
South Australia 5064

EEmail account:

pulse@cyburbia.net.au

**WARNING!!!**

If you ask me anything which is obviously contained within this document or the FAQ, I will reply with RTFM (Read the #\$\$@#\$\$ Manual). Sorry, but I hate idiots who can't be bothered to take their time to have a quick look through this document and write to me with stupid messages which have recently become a chore to answer (You'd understand if you received 4000+ EEmails...)

***BUT!***

If there *\*is\** something that you HAVE made an effort to understand/look for, then don't hesitate to write to me. The worst that can happen is that I reply with RTFM :)

\*\*\*\*\***ALSO**\*\*\*\*\*

If you have an incorrect reply-address, don't expect to receive anything.

If you are reporting a bug, let me know what version you are using.

If you find any major problem with the tracker, please try to find the circumstances which cause it to trigger - if you can't, don't worry - write to me anyway. It's just that it's much easier if I know a likely cause of the error. (Don't worry if it seems that the cause is very, very remote - I had a problem in my *GUS* playback routines which originally caused my palette configuration screen to hang - now that's obscure!)

Send money! I don't demand that you send me anything, (and I don't like spoiling a program by showing in reminder messages), but if you *\*DO\** use the program and think that it's worth contributing a small amount towards, then please do so... it would be GREATLY appreciated. Remember... even a few dollars will be appreciated! It's very unlikely that I'll make a huge profit (I doubt I could get close to sufficient money from this program to make it even worth \$1 for every hour I put in!). I won't be sending out any nice thankyou notes, or bound manuals... but I will include your names in the future versions of the tracker and I will EEmail you each new version the minute they are released if you wish.

I believe that software *\*SHOULD\** be cheap - you've spent perhaps a few thousand getting your computer, so you wouldn't want to spend TOO much more on programs! If you think this program is worth \$5, then send \$5. If you think it's worth \$10, then send \$10. If you think it's worth \$1,000,000,000... don't let me stop you :)

So... get together with a couple of friends and send me \$10... If you send me a significant amount, I'll put together a "special contributor's" list.. (Oeeerrrr :)

Methods of payment:

Bank cheque

Cash (Make sure it's securely wrapped!) - it doesn't have to be Australian currency, although it'd be nice if it could be!

Bank Transfer - EEmail me for my details

## **4. For the software programmers & hardware manufacturers**

### **4.1 Programmers**

The technical specs for the IT format can be found in the file, ITTECH.TXT, which should be in the archive in which you found this document.

If you decide to write a player, and need any help/clarification of the format, then don't hesitate to write to me!

### **4.2 Manufacturers**

At the moment, the hardware support for the tracker has been limited to:

- PC Speaker
- DAC on LPT1/LPT2
- Sound Blaster 1.xx
- Sound Blaster 2.xx
- Sound Blaster Pro
- Sound Blaster 16
- Sound Blaster AWE 32
- Gravis UltraSound
- Interwave
- Pro Audio Spectrum
- Pro Audio Spectrum 16
- Windows Sound System
- ESS ES1868 AudioDrive

If you would like me to support another soundcard (which isn't compatible with an existing driver, or if it has it's own, special native mode), then send a sound card AND a copy of the Software Developers Kit. I will \*try\* my best to incorporate them into the program if you do this (but there's the limitation of design difficulties - which shouldn't cause any problems in 99% of the cases.)

## **5. Closing Words**

I would like to thank the following people:

Psi: For introducing me to tracking with the brilliant *Scream Tracker 3*

Purple Motion: For being my music hero.... I still think "When the Heaven's Fall" is \*THE\* best S3M ever written.

Red Haze: For constantly bugging me to get the tracker done, and for his effort in testing out the program and making suggestions. (He's directly responsible for the inclusion of Ping Pong loops - I originally couldn't be bothered :) ) Also extensive testing of later versions of IT...

The WALKER: For his flow of ideas & beta testing and getting the PoP DiskMag finished.... tomorrow. Or is that next week? OK... so he has done it. Finally.... 6 months late. :)

Perception: For his constant support over the many years he ran his BBS

Psibelius: For being the nice friendly Epinicion founder that he is.

Dominic: For his thoroughness in beta testing the program, and providing suggestions. (He came up with more than double the number of suggestions than almost anyone else... even though many of them were not finally implemented :) )

Chris Jarvis: For his effort in beta testing the tracker and producing:

- Firestorm (released with v1.00 of the tracker)

- Pale Dreams! (released with v1.01 of the tracker)

- Firepower (released with v1.03 of the tracker)

Sidewalk (released with v1.05 of the tracker) and  
Fallen World (released with v2.01 of the tracker)  
These are ALL amazing pieces of work! (Yes, I'm jealous!)

Zilym Limms: For his technical help with the tracker and his love of Fast Tracker 2. Hahaha. :) Also for his information and source code for the PAS 16.

Benjamin Bruheim: For his super list of contributions to v1.01

ZaStaR: For his effort in putting together ITF - the font customiser! and ITTXT - the text file importer for IT's messages! Also for being the willing target of a barrage of beta versions, a great contributor of ideas, a musician with whom I could exchange work and for being a great dude all round!

Emmanuel Giasson: For creating the utilities MMCMP/MMUNCMP/MMTSR - the music module compressor!

Diablo: For creating the Impulse Tracker directories on ftp.cdrom.com!

Siren: For providing me a contact to obtain an Interwave card!

AMD: For providing me with an Interwave card to program for!

To the other beta testers:

Eric Bonython      Phorte  
Alistair Watts     Julian Ellis  
Alex Bates          David Rohrsheim  
Delta X              GD (Grave Digger)  
Andy Chen      ShawnM  
Clef                  Emmanuel Giasson  
KXMode

And anybody there who uses this program!!!

Big thanks and thumbs up to Advanced Gravis for releasing their Software Developer's Kit (SDK) free of charge!

Congratulations to Creative Labs for finally releasing their SDKs free of charge!

Especially \*BIG\* thanks to AMD for providing me with an Interwave card to use and the SDK!

Finally... I have a request.....

If \*YOU\* write something with this program... share it with the rest of the world!!!

If you have an iNet account, you can upload your songs to:

ftp.cdrom.com/pub/demos/incoming/music/songs/it

- remember to ZIP up your song, to use a \*lower\* case filename and to also upload a short description of your song in a .txt file, otherwise your file will just be deleted.

If you know of any other places where modules can be uploaded to, let me know and I'll include them in the above list.

## **6. How to get the latest version of Impulse Tracker**

There are several places around the internet where you can find the latest version of Impulse Tracker:

- 1) ftp.cdrom.com/pub/demos/incoming/music/programs } Check (1) first!
- 2) ftp.cdrom.com/pub/demos/music/programs/trackers
- 3) http://www.citenet.net/noise/it - American homepage
- 4) http://huizen.dds.nl/~im-pulse - European homepage
- 5) ftp://sdc.wtm.tudelft.nl/pub/music/utls/it - ftp site!

If you want to have the latest version sent to you via EMail, or at least be notified of new releases, then you'll have to send something to me :) Check section 3 on how to contact me.

## **7. Legal stuff**

No matter what happens, no matter how bad, I'm not going to be held responsible.

That's basically the same as any other license agreement, except you should be able to understand this one without an interpreter :)

YOU MAY NOT CHARGE ANYTHING FOR THIS PROGRAM - DISTRIBUTORS WHO ARE INTERESTED IN THIS PROGRAM MUST WRITE TO ME FIRST AND HAVE MY APPROVAL! (yeah, even by snail mail if you don't have EMail access).

USE OF THIS PROGRAM COMMERCIALY IS EXPRESSLY FORBIDDEN WITHOUT WRITTEN AND SIGNED APPROVAL FROM ME. COMMERCIAL USE INVOLVES ANY USE OF IMPULSE TRACKER IN WHICH MONEY IS INVOLVED (SPECIFICALLY THE CREATION OF ANY MUSIC WHERE PAYMENT IS INVOLVED). THIS PROGRAM IS ONLY FREEWARE FOR NON-COMMERCIAL USE.