Settings Menu

The **Settings menu** is used to configure your Soundscape Hard Disk Recorder and allows you to select the following functions.

Active Tracks per Unit

Active Tracks per Unit determines the number of active tracks per unit as 8, 10 or 12. The setting is also shown in the status bar as T/U. If you select 12 tracks per unit and a 'DSP processing power limit reached' error message appears, then try reducing the number of tracks per unit to 10, as this is related to the drive performance. If you use a number of simultaneous fades in your arrangement, then you could also try checking the same arrangement with fades turned off.

If there are more active tracks in the arrangement than the active tracks per unit setting, the parts that cannot be played will be displayed with a large X'. The amount of tracks available for recording and playback may be different due to the speed of your hard disk. (See the "tested drives" chart in the **Troubleshooting chapter**.)

Master Clock / Input

The Master Clock/Input parameter sets the audio input source and the source of the **sample rate clock** or **'Master Clock'**.
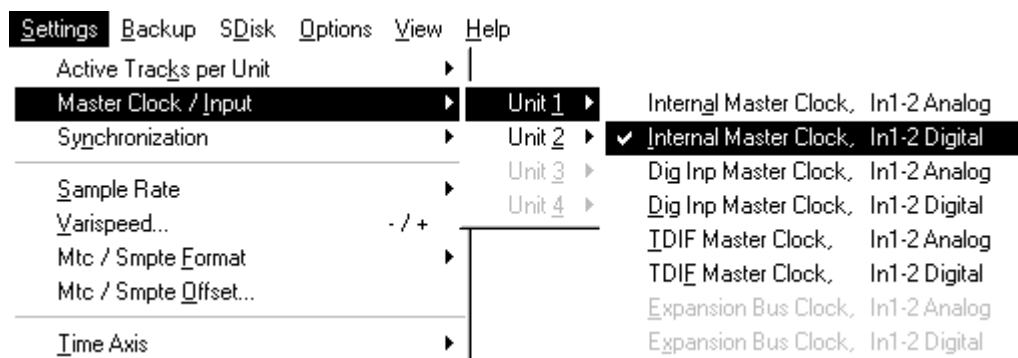
Internal Master Clock, In1-2 Analog selects the analogue inputs for recording and uses Soundscape's internal 'Master Clock' to set the sample rate.

Internal Master Clock, In1-2 Digital selects the digital inputs for recording and uses Soundscape's internal 'Master Clock' to set the sample rate.

Dig Inp Master Clock, In1-2 Analog selects the analogue inputs for recording and uses the 'Master Clock' provided by the external source (*i.e. DAT, CD or digital mixing desk*).

Dig Inp Master Clock, In1-2 Digital selects the digital inputs for recording and uses the 'Master Clock' provided by the external source (*i.e. DAT, CD or digital mixing desk*).

TDIF Master Clock, In1-2 Analog selects the analogue inputs for recording and uses the 'Master Clock' provided by the external TDIF source (*i.e. SS8IO-1, DA38, or digital mixing desk*).

TDIF Master Clock, In1-2 Digital selects the digital inputs for recording and uses the 'Master Clock' provided by the external TDIF source (*i.e. SS8IO-1, DA38, or digital mixing desk*).

Note: *When using TDIF Master Clock make sure the desired sample rate is set for Soundscape before switching to TDIF input.*

Expansion Bus Clock, In 1-2 Analog selection is only available when the software detects a connection during bootup. See **Expansion Bus Clock Input** later in the chapter.

Expansion Bus Clock, In 1-2 Digital selection is only available when the software detects a connection during bootup. See **Expansion Bus Clock Input** later in the chapter.

When to use "Internal" or "External" modes

Internal Master Clock, In1-2 Analog

When you want to record analogue signals into Soundscape at your chosen sample rate.

When you are using the unit as the "master" in a multiple Soundscape setup.

Internal Master Clock, In1-2 Digital

When you have a digital effects unit with digital inputs and outputs connected to Soundscape in a "loop". Soundscape can supply the sample clock reference, and therefore does not need to lock to a different sample clock at it's input.

When you *do not want* to use the sample rate clocks from an external source.

Dig Inp Master Clock, In1-2 Analog

When you are recording an analogue source while using an external digital effects unit or other digital device that has digital outputs plugged into Soundscape.

Dig Inp Master Clock, In1-2 Digital

Whenever you are recording from a DAT or other digital device. Soundscape will automatically synchronise to the sample rate received at it's digital inputs.

All 'slave' units in a multiple Soundscape setup, so that they are synchronised to the master unit sample rate.

TDIF Master Clock, In1-2 Analog

When you wish to record an analogue source through inputs 1-2 while using a TDIF device such as the SS8IO-1, Tascam DA88 or Yamaha Pro-3.

TDIF Master Clock, In1-2 Digital

When you wish to record a digital source throught inputs 1-2 while using a TDIF device such as the SS8IO-1, Tascam DA88 or Yamaha Pro-3. Soundscape will automatically synchronise to the sample rate received at it's digital inputs.

All 'slave' units in a multiple Soundscape setup, so that they are synchronised to the master unit sample rate.

If you have selected an external source sending to your Digital Inputs, you will not be able to select a different sample rate.

If there is no digital signal present at Soundscape's digital input when you select **Dig Inp Master Clock**, after a few seconds the following dialogue box will be displayed:-
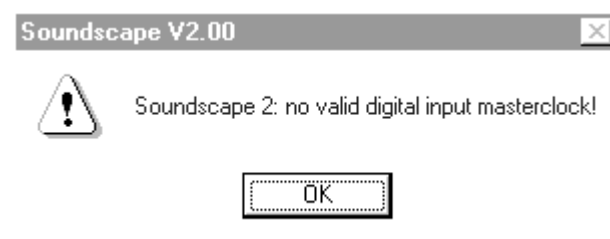
If this persists after you have clicked on **OK** or pressed the **Enter key**, check that your DAT recorder is set to use internal sample rate, or check your connections.

Using TDIF Master Clock

Selecting TDIF Master Clock means that the system gets the sample rate directly from the TDIF port, so that the external device is the 'clock master'. This is the setting that should be used when connecting to a Tascam DA-38, DA-88 or DA-98 and can also be used with a Yamaha digital console.

The Soundscape SS8IO-1 can be used with the TDIF Master Clock setting if you select Internal Clock for the SS8IO-1, or if the ADAT or External clock settings are used for the SS8IO-1 and the sample rate is then provided by another device.

Note: *Make sure that you select the correct internal sample rate before switching to TDIF Master Clock, as the SSHDR1-Plus does not automatically detect whether the external TDIF signal is at 44.1kHz or 48kHz. If the internal sample rate is incorrect the timecode calculations used in the software will also be incorrect.*

Most problems with transferring multi-channel digital audio signals arise from incorrectly routed wordclock signals or clock settings.

Using a TDIF Connection between two SSHDR1-Plus Units

You can use the TDIF port as an 8 channel bi-directional 'bus' between two Soundscape SSHDR1-Plus units, so that signals can be routed from one unit to another. For example, you could use real-time effects that are running on one unit with tracks that are played from the other.

Select Internal clock for the first unit and TDIF Master Clock for the second. The second unit then locks to the sample rate from the first unit and if you also connect the S/PDIF or AES/EBU signals between the units, these will also lock correctly. Use the TDIF routing (*e.g.* IN TD1-2 U1) in the V2 Mixer for the inputs, outputs and send elements to connect the signals where required.

Note: *If the second unit is set to use Digital Input Master Clock and there is an S/PDIF or AES/EBU signal from unit 1 to unit 2, the TDIF may not be fully locked, as the phase may be incorrect. Switch first to TDIF Master Clock and then back to Digital Input Master Clock to ensure correct phase between the units.*

Using Chase Slave mode with TDIF

In Chase Slave mode, the internal SSHDR1-Plus sample rate is varied in order to match the incoming timecode rate. To use the TDIF connections, the external unit must use the TDIF signal to provide the masterclock. With an SS8IO-1, use the TDIF Clock setting.

For a Yamaha 02R or 03D mixing console, set the wordclock source to the TDIF port. The Tascam DA-38, DA-88 and DA-98 cannot use their external TDIF port to provide the sample rate, but they can use an external wordclock signal. In this case, you need to use an external unit (like the Rosendahl WIF unit supplied by Soundscape Digital Technology) which converts the LTC or VITC timecode rate into a wordclock signal that matches the timecode. Connect the wordclock output of the convertor to the Tascam's wordclock input and select external wordclock. The SSHDR1-Plus must then use TDIF Master Clock mode with Trigger Slave mode.

With a suitable cable, a TDIF port can be split into separate 8 channel input and output connections.

Expansion Bus - Using Superclock Signals & Chase Slave

Many systems commonly use a wordclock signal to synchronise different digital devices for passing digital audio between them. Wordclock signals are at the sample rate frequency of the system, but digital components like AD/DA converters need much higher frequency clock rates to synchronise the digital audio streams. These higher frequency clocks are generally known as 'Superclock' or '256 x Superclock' and for 48kHz, the superclock frequency is 12.288MHz, 256 times the sample rate. Most digital devices use a Phase Locked Loop (PLL) circuit to generate the superclock from the wordclock signal, but these PLL circuits all have different locking delays. This causes problems during chase slave, when the wordclock signal is derived from a varying timecode (*i.e.* with tape stretch or when starting a VTR etc.) and there are two digital devices individually generating a superclock.

During the lockup period, the different locking delays can mean that the two superclock signals are not exactly the same frequency or phase and so digital clicks are produced in the digital signals being transferred between the digital devices. This depends upon the amount or rate of change of the wordclock signal and the locking time of the PLL circuitry. The only way to prevent this is also to transfer the superclock signal between different devices, so that only one PLL is active for the system. Many systems need this solution, but not all have the capability to use external superclock signals.

The SSHDR1-Plus includes a rear panel Expansion Bus which contains bi-directional superclock and L/R clock (like wordclock) signal connections and a 512 channel Time Division Multiplexed (TDM) audio bus. The SS8IO-1 can also generate and receive a superclock signal (set with the rear panel Option switch), so that both units can be locked together via the superclock signal if required. If you experience digital clicks when using the SS8IO-1 with the SSHDR1-Plus in Chase Slave mode, then this is the best solution.

With two SSHDR1-Plus units connected via TDIF, however, in Chase Slave mode the PLL for the digital input (S/PDIF or AES/EBU) will lock more quickly than the PLL for the TDIF port. You may find that using Digital Input Master Clock instead of TDIF Master Clock for the second unit may help - select TDIF first, then select the Digital Input. Make sure that you connect the digital S/PDIF signal between the two units as well as the TDIF connection. This may not work in all circumstances though, so the best solution may still be to use the Expansion Bus and superclock signals.

Expansion Bus Clock Input

In the Settings menu, under Input, there are two options for using Expansion Bus Clock. As the superclock and L/R clock signals are bi-directional (*i.e.* software switched as input or ouput) the options are not available if there are no signals detected when the software is started. In this case, both signals are set as outputs and the superclock can be used to connect to the SS8IO-1 superclock/wordclock input.

Multiple SSHDR1-Plus units can be synchronised using the Expansion Bus to phase lock the entire system, also for transferring multiple TDIF signals to other devices like digital consoles or digital multitrack tape machines. The procedure is that all the Expansion connectors are linked, and when the software is started, the superclock & L/R clock inputs for each unit are checked in turn and those receiving a signal are switched to input mode. The normal situation would be that Unit 1 provides the masterclock for the entire system. Each of the other 'slave' units should be set to Expansion Bus Clock mode.

Synchronisation

Soundscape can synchronise to external devices as a 'master' or 'slave' using it's MIDI IN or MIDI OUT connections.

Select **MTC Master** (**M key**) if you want to use Soundscape as the master unit, and a slave device is synchronised using MIDI Timecode. When your Soundscape is in PLAY or RECORD modes, MIDI Timecode will be generated from the MIDI OUT socket.

Select **MIDI Clock Master** if Soundscape is to be the master unit and a slave device such as a MIDI sequencer is synchronised using MIDI Song Position Pointer and MIDI Clock Messages. In PLAY or RECORD modes, Soundscape will generate MIDI SPP and Clock messages from the MIDI OUT socket, the actual MIDI Clock timing depending upon the tempo value selected.

Select **MTC Chase Slave** (**S key**) mode if Soundscape is to be synchronised as a slave, to an external master device which uses MIDI Timecode. In PLAY or RECORD modes, Soundscape will then wait for MIDI Timecode messages at it's MIDI IN socket and will lock to the timecode in about 1 second.

⚠ **With Mtc Chase Slave mode, the internal sample rate is determined by the incoming timecode rate. This means that digital inputs cannot be used, as then the internal sample rate would need to be locked to the sample rate received at the digital input (except for the Dig In, Internal M.C. mode, where the sample rate is provided by Soundscape).**

Soundscape implements a full **'Chase Lock'**, which means that it continuously monitors the timecode, and if the incoming timing changes, Soundscape will vary it's sample rate to maintain optimum synchronisation. This is essential if synchronising to a tape machine which has timecode recorded on one track, as the tape speed can vary and without chase lock significant timing errors would occur. Soundscape will also follow large changes in timing which occur when using varispeed etc.

⚠ **If you are RECORDING music performances while "CHASING" a tape machine or other source that may produce large changes in timing, you are likely to hear pitch changes on playback when using a stable Mtc source (MIDI sequencer, DAT etc...) or Soundscape's internal clock.**

With **MTC Trigger Slave** mode, the received MIDI timecode only provides the **start** and **stop** location and the sample rate used can either be internal, or derived from the digital input as required. This means that with external digital equipment which provides both a timecode and a digital audio signal (such as a multi-track digital tape recorder), full lock-up is provided by the digital input signal once MTC has triggered playback or record.

To use SMPTE timecode with Soundscape requires an external MIDI-SMPTE Converter which can convert SMPTE into MIDI Timecode and vice versa. Most low cost converters can read and write **LTC** and there are several which can read and write **VITC**.

VITC and MTC Full Frame - SS Slave Application

MTC Full Frame is a message that is transmitted by VITC-MTC converters (such as the Rosendahl VIF or WIF units supplied by Soundscape Digital Technology) during pause or slow running (jog/shuttle). Start the **SS Slave application** from the Start Menu, so that when a Full Frame message is received by the PC's MIDI input the play pointer is positioned at the received timecode. This ensures that it is always at the frame shown on the VTR, ready for an edit to be made or a sound effect to be placed. To use a VITC-MTC converter, connect the MTC output into the SSHDR1 MIDI IN (for chase slave to normal MTC Quarter Frame messages) and then connect MIDI THRU into the PC's MIDI input.

Sample Rate

Soundscape supports all standard sample rates commonly in use.

Most types of consumer equipment (CD players) sample at 44.1kHz (44100Hz) and this is adequate for high quality recording in most instances. Professional or studio equipment often uses the highest quality 48kHz sample rate, or can be switched between 44.1kHz and 48kHz. Broadcast equipment may use 32kHz, as the audio bandwidth requirement is lower for radio or TV. 22.05kHz will provide adequate bandwidth for speech or lower bandwidth signals. 44056kHZ and 47952kHz are provided for synchronising to the "drop frame" SMPTE used with NTSC video.

With higher sample rates, more disk space is required to store the digital audio, so it makes sense to use the lowest sample rate which will provide adequate reproduction of your material. The table below gives a guide to the amount of disk space required per minute of track storage (mono 16 bit) at the different sample rates. For example, to work out the disk size required to store 5 minutes of 4 track audio at 44.1kHz, multiply the figure for 44.1kHz by 5, then by 4, giving 101MB.

Soundscape

SAMPLE RATE	Megabytes per Track Minute (16 bit)
22.05 kHz	2.52 MBytes
32 kHz	3.66 MBytes
44.056 kHz	5.04 Mbytes
44.1 kHz	5.05 MBytes
47.952 kHz	5.48 Mbytes
48 kHz	5.49 Mbytes

Varispeed...

The **Varispeed** setting provides a +/-10% variation in the sample rate. Varispeed can be set either using the - and + keys (*from the main key area, **not** the numeric keypad*), which change the varispeed value by 0.1% each press, or by selecting **Varispeed** from the **Settings menu** and entering the amount required. The **Varispeed** parameter is saved in the arrange file.
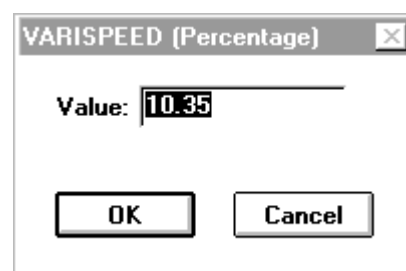
⚠ **Varispeed can be active in RECORD mode. Therefore, if you save an arrangement with a Varispeed value, your new recordings will be made at the wrong speed! If the Varispeed value is not 0.00%, the Sample Rate displayed in the Status bar will be shown in RED!**

Mtc / SMPTE Format

Soundscape can work with any of the commonly used SMPTE or EBU frame rates, 24 frames per second for film, 25, 29.97, 29.97df, 30 or 30df frames per second for video.

NTSC Video

Drop frame timecode (29.97df and 30df) is fully supported in **Master** and **Slave** modes. When the **Time Axis** is set to SMPTE time, the time values displayed the drop frame specification, so that 2 frames are dropped at the start of every minute, except for 0, 10, 20, 30, 40 and 50 minutes during each hour.

In total 108 frames are dropped each hour, the difference between 29.97 frames per second and 30 frames per second. This provides a convenient way to deal with seconds and minutes at 30 frames per second, instead of using fractions of seconds and minutes at 29.97 frames per second.

MTC / SMPTE Offset

The **MIDI Timecode/SMPTE Offset** is used to align the Soundscape correctly with the timecode from an external device.

For instance, if a section of film which requires an audio track has timecode starting at 02:05:20:00 (2 hours, 5 minutes, 20 seconds) entering this value for the **Offset** will make the arrangement also start at this time. In **'Slave'** mode, the **Offset** is subtracted from the external timecode and in **'Master'** mode it is added to Soundscape's timing before the MIDI Timecode is generated.

The **Mtc/Smpte Offset** allows a maximum limit of 24 hours.

Time Axis

Timing can be displayed in the **Time Axis Bar** in the **Arrange Window** as a measure (*i.e. Bars, Beats & Ticks)* or in SMPTE timecode form. When 'Bars' is selected, the tempo and time signature settings determine the actual relationship between real-time and the measure value displayed.

If you are synchronising to a MIDI sequencer, a measure display is probably more useful, as both applications will then display the same value, as long as the time signature and tempo values are identical.

SMPTE display is best used for "audio for video" or radio productions.

Waveform Axis

The display of waveforms and fade curves can be changed from logarithmic to linear using the **Waveform Axis** entry. The logarithmic mode has the effect of compressing the waveform, making lower levels more visible. It also makes a linear fade appear curved, due to the compression.

▶ *Linear Waveform Axis*

▶ *Logarithmic Waveform Axis*
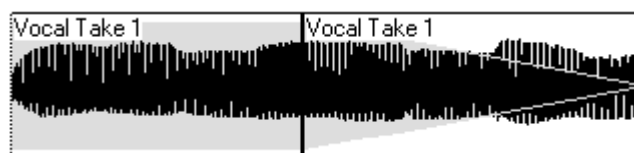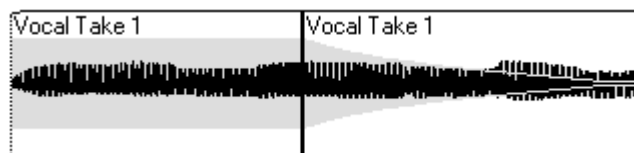
Snap

<u>O</u>ff
✓ <u>B</u>ar
1/<u>4</u>
1/<u>8</u>
1/<u>1</u>6
1/<u>3</u>2
1/4 T
1/8 T
1/16 T
1/32 T
<u>S</u>econd
<u>H</u>alf Sec
<u>F</u>rame
<u>Q</u>Frame
10<u>t</u>hFrm
1<u>0</u>0thFrm

The **Snap** value is used when making edits with the **cut tool**, **copying** or **moving 'parts'** around the **Arrange window**, setting the left and right locators or when selecting the song position with the play pointer.

If a single part is positioned with the mouse using the **move** or **copy tools**, the start of the part will locate at the nearest bar, beat, 8th, 16th. second or frame depending upon the current snap value. If the snap is turned off, the part can be positioned with sample accuracy (*i.e..* the part can start at any sample in the arrangement).

If a group of parts are selected for moving or copying, the part which is being pointed at with the mouse when the group is dragged will be 'snapped' to the current snap setting. The other parts of the group will follow the moved part, keeping their original relative positions.

The Snap function is not the same as the quantise function used by MIDI sequencers.

In a sequencer MIDI notes can be moved around easily, as they are just represented by numbers in the computer's memory. To perform the same operation automatically with digital audio would require individual notes to be identified in the waveform, cut up and then moved around without causing any noise or glitches, a process which is currently beyond all but the most powerful systems.

However, Soundscape does allow you to to cut up the waveform into smaller parts and manually move the sections using a new snap value.

If you really want to quantise your drummer, use **compute tempo** to find the nearest BPM - use the **noise gate** to cut up the drum performance - set the snap to 16ths or 32nds and manually move all the beats into perfect time. *Note: this also works for sloppy bass, vocals, guitar etc...*

Record Setup...

The **Record Setup** dialogue box allows you to set Pre/Postroll durations, how the system allocates SDisk space for manual punch outs and selects how Soundscape responds when making a looped recording.

Preroll

Active - Soundscape will play from a preset time before the left locator (using the value set for **Length**) and will drop into record mode as normal when the left locator is reached.

Preroll is also used in Looped recording mode. Instead of returning to the left locator when looping, Soundscape will start playing from the Preroll length before the left locator allowing you to prepare for the next recording attempt.

Length - This value sets the loop position before the left locator and can be set in bars, beats and ticks or in SMPTE time. *e.g. If 1 bar is selected, the loop will start 1 bar before the left locator.*

Postroll

Active - Soundscape will drop out of record when the right locator is reached and carry on playing for the duration of the postroll length before returning to the preroll position in looped recordings. Postroll is particularly useful in dialogue replacement. It is much easier for the V/O artist to determine the "feel" or "rhythm" of the dialogue if the playback carries on past the replaced section.

Maximum record time per Track for manual punch out

It is possible to allocate a set duration of disk space per track for "manual punch out". Alternatively, a setting of **"0"** allocates the maximum amount of free contiguous disk space.

Autostop

At end of recording - When selected, Soundscape drops out of record and stops when reaching the right locator. When unselected, Soundscape drops out of record at the right locator and carries on playing until you hit stop.
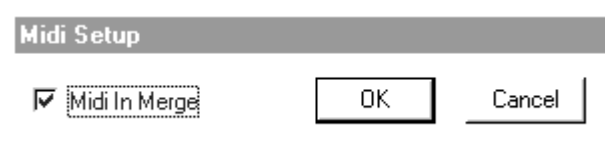
x

x

.

x

.

Colors

You can set a different colour for each physical track output in play or record modes. This allows different tracks to be identified easily when multiple rack units are used.

Time Axis ▶		Rec Out1
Waveform Axis ▶		Rec Out2
Snap ▶		Rec Out3
		Rec Out4
Record Setup...		Rec Out5
Midi Setup... ▶		Rec Out6
		Rec Out7
Muted Parts / Takes Solo Output ▶		Rec Out8
Colors ▶	**Unit 1 ▶**	Rec Out9
Customize Tool Page...	Unit 2 ▶	Rec Out10
Preferences ▶	Unit 3 ▶	Rec Out11
SDisk Auto Calibration ▶	Unit 4 ▶	Rec Out12
SFile Manager sort ▶		
Marker Directory sort ▶		**Play Out1**
Big Current Time ▶		Play Out2
AVI Player ▶		Play Out3
Mix Window ▶		Play Out4
		Play Out5
Save Settings		

You will be presented with a standard Windows colour palette to design your own colour scheme.
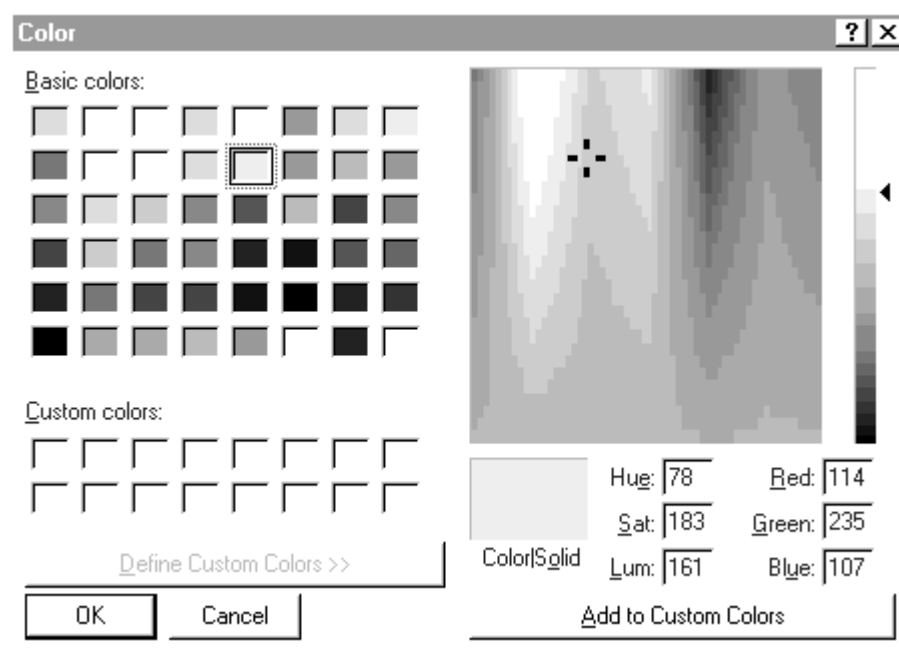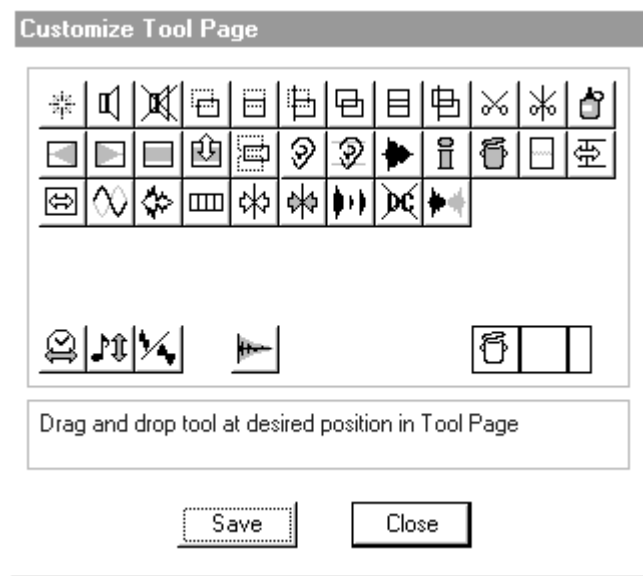
Soundscape

Customise Tool Page...

You can access the **Customize Tool Page** pop up window to alter the currently selected tool page. It can also be activated by double clicking on a free space on the toolbar.

(see **Customisable Toolbar** in the **Tutorial chapter**)

Preferences

Fast Redraw

Selecting **Fast Redraw** mode allows Soundscape to take full advantage of the improved screen update speed which is possible with a graphics accelerator card.

Fill Waveform

Selecting **Fill Waveform** displays waveforms as solid black. Disabling **Fill Waveform** displays waveforms as outlines and speeds up redraws.

Show Part Name in waveform

This setting selects whether all names are visible when parts are displayed as waveforms as shown above.

Show Part Volume

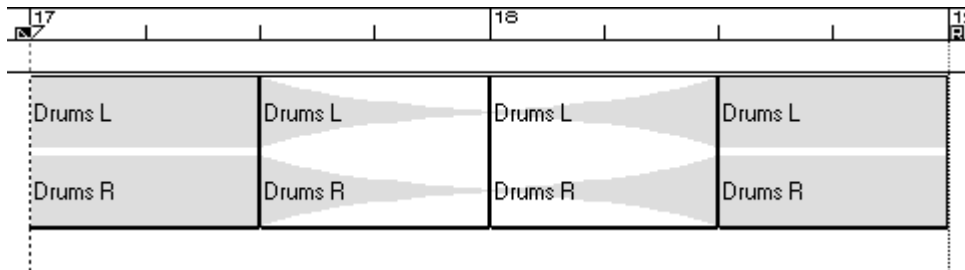When **Show Part Volume** is enabled, you can view the volume contour (shown by the part colour). On some slower PCs (especially 486SX or 386SX/DX without a 387 Co-processor) the display of volume contours may make screen redraws slower. With **Show Part Volumes** off, the contour is always shown for selected parts or when dragging the contour with the **Volume Trim tool**.


Auto Delete Takes

If **Auto Delete Takes** is selected, the take will be deleted without displaying any warning dialogue boxes.

⚠️ **Be careful when using Auto Delete Takes, as you can only UNDO the last TEN operations and in certain circumstances (ie. if the disk has been overwritten by another operation) then it may not be possible to undo the deletion.**

Confirm Deletes

This setting enables an extra warning when a delete part operation is performed. The following dialogue box is now displayed each time.
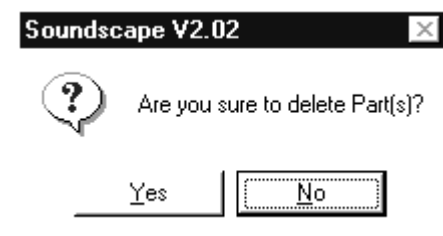
Deglitch

The **Automatic De-glitching** can be turned ON or OFF. Automatic deglitching mode detects when the audio material is continuous before and after the cut when a part is fading in or out, and then does not perform a deglitch. See **Joining Parts Together** in the **Tutorial chapter**, for more information on the de-glitching function.

Keep Part Selection

When selected, the **Keep Part Selection** parameter allows the currently selected parts to remain selected after an edit. This is useful if a number of different operations have to be performed on a group or if all selected parts are not on screen at the current zoom level and to reselect them would mean zooming out, then zooming in again etc...

Keep all generated Waveform files

When selected, all **Waveform files** generated will automatically be saved in the c:\Soundscape\wvf folder when you exit the Soundscape software. If not selected, only the waveform files for takes that are still on the SDisk will be saved and all others will be deleted from the PC.

Play till next Marker

When **Play till next Marker** is enabled, Soundscape will stop playing at each *marker*. This is useful for "previewing" edits, or even "cueing" up audio for radio spots or theatre. Putting *markers* at the beginning of each new "cue" would define the start and stop points of each audio segment. So, when the audio is triggered, it plays "till the next marker" and is automatically *cued* for the next trigger.

Fast REW/FF Mode

Select Fast REW/FF mode to set the speed 4 times as fast as normal.

Remote Control Enable

This setting enables the SSHDR-1 to be controlled by the JL Cooper CS10, MCS or the Penny+Giles MM16/DC16 software modules.

Shift Markers with Global Insert/Delete

When active this setting maintains the relationship between the markers and the arrangement when performing a global insert or delete. If unselected, any global inserts or deletes will not move the marker positions.

Digital Transmit Channel Status

This allows **Consumer** or **Professional Channel Status** to be selected for the digital data stream transmitted from the digital outputs. Some professional DAT recorders will not record from their digital inputs if **consumer** format is received, so set professional status if you are using these machines and you have problems when set to **consumer** status.

Soundscape responds to either format automatically, so there is no need to set the format for receiving a digital input.

If **Digital Input Masterclock** is selected, but received Channel Status sample rate information is 'not indicated' (*e.g.* this is the case for the OTARI DTR900), then Soundscape maintains the current sample rate selection.

SDisk Auto Calibration (*Used for older drives which use a periodic thermal calibration*)

```
Preferences                    ▶
SDisk Auto Calibration         ▶   ✔  Optimize SDisk Auto Calibration
Parametric EQ                  ▶      Recalibration Duration...
Take Directory sort            ▶      Recalibration Timeout...
Marker Directory sort          ▶
```

Optimize SDisk Auto Cal

When the **Optimize SDisk Auto Calibration** mode is selected, you can determine the time between automatic recalibrations. Soundscape performs these recalibrations just before starting record or play, in order to maximise uninterrupted disk operation.
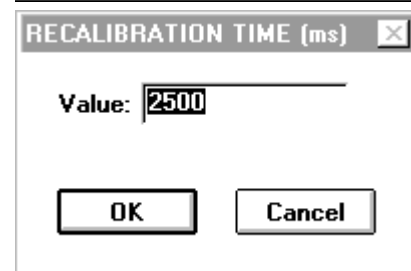
Recalibration Duration...

The default setting is 120secs, but you may wish to increase this to reduce the frequency of the delay while recalibration occurs.
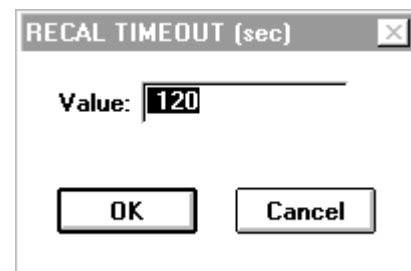
Recalibration Timeout...

The **Recalibration Timeout** value sets the length of time that the hard disk is given to recalibrate. The default setting is 2500 ms, but you may wish to decrease or increase this depending on the type of hard disk.

RECAL TIMEOUT (sec)
Value: 120
OK   Cancel

RECALIBRATION TIME (ms)
Value: 2500
OK   Cancel

Note: This mode is not required for the hard disks that we recommend for use with Soundscape. However, you may have a disk in use that is not recommended, but apart from the Periodic Thermal Calibration cycles that the disk performs regularly (about once every 30 minutes), the disk works fine. This mode will optimise the amount of uninterrupted time for which you can continuously record or playback.

SFile Manager Sort

The four **SFile Manager Sort** modes are:-

by Name	- in alphabetical order
by Recording Date & Time	- in date/time order, oldest last
by Length	- smallest first
by SoundDisk position	- in order of storage on disk

Marker Directory Sort

The three **Marker Directory Sort** modes are:-

| by **ID** | - | in order of Marker ID numbers |
| by **Name** | - | in alphabetical order, last |
| by **Loc** | - | in order of marker location |

Big Current Time

It is possible choose the font, size and colour of the **Big Current Time** display. It will display Bars+Beats or SMPTE depending what you have selected in the Time Axis display setting under the **Settings menu**.

Selecting **Font** will open the following settings window:-
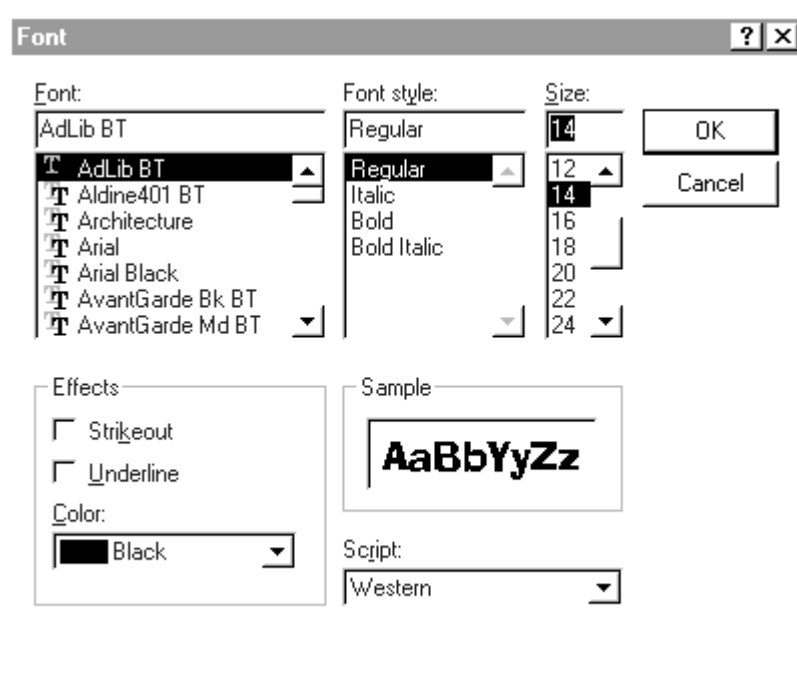
Here you can choose any Windows 'True Type' font for your **Big Current Time**.
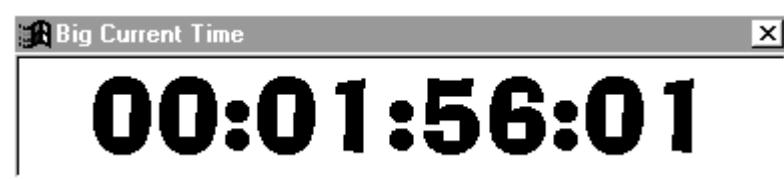
Selecting **Background Color** will open the standard Windows colour pallette for you to choose your own background.

Selecting **Always on Top** will keep the Big Current Time visible even if you switch to another programme such as a MIDI sequencer.

Note: The smallest sizing of the time code reader is determined by the font and size.

AVI Player

AVI Player ▶ | Offset
Mix Window ▶ | Display Size ▶ ✓ Auto
Save Settings | Always on top | 1/4
| | 1/2
| | 1/1
| | 2/1

AVI Files

AVI files are opened via the "Open PC file AVI" selection under the **File menu**.

Offset

The first option available is to set the **AVI timecode offset**. The easiest way to do this is to make sure that the **SMPTE Offset** in the **Settings menu** is set to **00:00:00:00**. This will ensure that the first frame in the AVI file will be displayed. If you have captured with timecode in vision, you can then just read this from the video window and type this in as the **AVI Timecode Offset**. If not, you may have to identify a significant event in the video that can be related to the audio in some way.

In any event, you must ensure that the SMPTE Format (*i.e.* 25fps) matches the timecode format of the captured video. To check that there is synchronisation from the start to the end of the video file, move the Soundscape play locator and check that the video shows the same timecode as the Soundscape transport window.

AVI Offset (SMPTE)

10:00:00:00

Save to AVI File

OK   Cancel
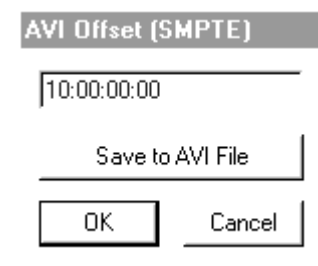
Save to AVI File allows you to save the offset within the AVI file so that the next time you load the file, the offset will be automatically set.

With AVI files captured at a lower frame rate than the current Soundscape SMPTE Format, you will find that as the playback of Soundscape advances, there will be missing frames in the AVI file, so the same frame is displayed twice. However, there should be no overall synchronisation error between video and audio. If you do find a slippage, then the AVI file may have been captured with dropped frames.

⚠️ **You may find that there is a one frame error right at the start of the file. This is common with several of the capture cards we have tested, but all it means is that the AVI timecode offset must be adjusted one frame up or down.**

Display Size

The second option is setting the display size. If set to **Auto**, the file will be displayed at the size and frame rate it has been captured at. So if the AVI was captured at 320x240 it will occupy 320 horizontal pixels and 240 vertical. This may take up too much space on your Soundscape window, so you have other settings to make the display 1/4, 1/2 ,1/1 or 2/1 of the original captured size.

Always on Top once again allows the video file to be displayed in front of all other active Windows programmes.

About AVI Files and Video on the PC

An AVI file (Audio Video Interleaved) is a standardised format file for video and audio capture and playback on the PC. With **Microsoft Video for Windows**, different frame rates, image sizes, video compression standards and colour formats are supported. Software only decompression algorithms are installed automatically in Windows 95/98/NT4™, but these require a very fast PC even to playback reduced frame rates of say, 15fps (frames per second) with a small size image. For full screen video at full frame rates (*i.e.. 25 or 30 fps),* hardware decompression cards are required and usually a fast PCI bus PC with a fast hard drive is also necessary.

The Soundscape AVI Player receives timecode from the Soundscape SSHDR1 hardware and uses Video for Windows to display the frame required to synchronise to the audio.

Scrubbing audio and video is always frame accurate and every frame available in the AVI file will be played even if full rate playback is not possible due to PC speed limitations. This makes it easy to accurately place sound effects for AV post-production work. If the PC is not fast enough to play every frame requested by the AVI Player at normal play speeds, then there will be no slip between the video and audio tracks, as the next video frame available will be displayed in sync.

You may find that Microsoft Media Player can play a file smoothly, but played using the Soundscape AVI Player it is jerky or cannot be played at full rate. This is because Media Player uses a simple playback mechanism, one frame after another, without regard for any synchronisation to audio rates, whereas the Soundscape AVI Player must maintain synchronisation, and so requests that Video for Windows locates each particular frame in the file as necessary. This takes more processing time from the PC and so you may need to optimise playback by reducing the image size or frame rate. You should also defragment the disk to ensure that unnecessary disk head movement does not interrupt playback. You may find that you need to upgrade your PC in order to get optimum playback performance. The same is true even if you have a video capture/playback card installed.

There is also a **Command Line Switch (-sx)** that allows optimisation of the AVI update rate to obtain smoother playback with faster PCs. See the Command Line Switches section in the Installation chapter for details.

Capturing the Video

For full motion (up to 60 fields per second) full screen video capture, you will need a PCI bus Pentium PC (166 MHz or faster), a minimum of 16MB of RAM, a separate, fast IDE hard drive (SCSI is sometimes recommended) and a Bus Mastering PCI video capture card (*e.g.* Fast AV Master). Most capture cards

now provide Video for Windows drivers and so can work with the Soundscape AVI Player. Usually a capture utility is provided with the card, but you can use Microsoft's Vidcap or a video editing application like Adobe Premiere. In any case, you should endeavour to capture the video without dropped frames.

We recommend that you capture the video in "single field" with no audio.

You do not have to capture the audio (most capture utilities allow you to deselect audio recording) unless you plan to save it as a .WAV file for import into Soundscape. This will also reduce the amount of data that your PC hard drive has to store per second. 16 bit stereo audio at 44.1kHz has a data rate of 176.4 kilobytes per second in addition to the compressed video data (150kbytes per second to several MB per second) and this could cause dropped video frames. Most video editing applications can save the audio from an AVI file as a .WAV file and can also import a .WAV file (exported from Soundscape) and ocmbine it with a video sequence for layback to a VTR.

Lower cost ISA bus cards can also be used, but do not expect to be able to capture a broadcast quality image! However, they will be fine for recording a smaller size/resolution image in a video overlay window on the PC monitor that can be used with the SSHDR1 for AV post-production work.

The AVI file format supports a timecode offset for the first frame of the video, but as most capture cards do not read timecode (LTC or VITC) they are unable to record video against timecode. The Soundscape AVI Player allows you to enter a timecode offset and to write it to the AVI file if required. To make this easy to determine and enter, it is recommended that for AV post-production, you capture the video with BITC (burnt-in timecode, *i.e.*. timecode in vision). Most professional video players can provide this, but you can also obtain low cost synchroniser units that can add this to the video signal before capture (*e.g.* Midiman Video Syncman LTC/MTC/VITC/BITC).

You can select parameters in the capture utility which optimise the video recording. Some of these affect the output rate to hard disk (compression ratio, frame rate, image size, colour resolution), but others affect the amount of work that the compression device on the card must perform per frame (*e.g.* number of key frames, compression type). You may find that in order to capture without dropped frames, you have to reduce the frame rate to say, 20 frames per second. This will still be OK for synchronisation to speech (lipsync). A basic rule is that if you cannot play the file properly using Media Player, you will never be able to use the Soundscape AVI Player.

Tested Video Capture Cards

FAST Electronics AV Master (PCI bus, full motion, full screen, separate monitor, SVHS quality output) - *Recommended*

FAST Electronics DV Master (FireWire In/Out - Requires a fast PC) - *Recommended*

Soundscape

MIRO DC30 (PCI bus, full motion, full screen, separate monitor, SVHS quality output) - *Recommended*

Other cards that work with Video for Windows (*e.g.* Truevision Targa 1000/2000, Perception Video Recorder) will usually work fine.

Contact your local Soundscape distributor or supplier for a list of tested cards.

Save Settings

You can save the current values from the **Settings menu** to the initialisation (.ini) file using **Save Settings**. Also saved are the currently selected tools, the vertical and horizontal zoom levels, the screen positions, sizes and orientation of all windows. The **Record Track Arming** and **Record Track Positions** are not saved in the initialisation file, but are saved in the arrange file (.arr), with the **Sample Rate**, **Varispeed**, **Mtc/SMPTE Format**, **Mtc/SMPTE Offset**, **EQ**, **Tempo** and **Time Signature**.

When multiple Soundscape SSHDR1-Plus units are being used, the settings function will be saved for all units automatically when a **Save Settings** is performed.