Pristine Sounds 2000
User guide

This userguide covers both Pristine Sounds 2000 Light and Pristine Sounds 2000 Pro. The differences between the two versions are stated functionwise.

---

alien connections

February 7, 2000

What is PS 2000?

Pristine Sounds is a digital waveform editor. It can be used for editing already existing waveforms or to record new ones. Editing features include suppressing noise from analog equipment and clicks & pops from vinyl records. All common kinds of functions like filtering, reverberation, compression etc. are also available.

PS 2000 can work with up to 64 individual waveforms at a time, each hosted in a unique MDI (Multi document interface) window

Each waveform can be up to 2Gb (Gigabyte) big. An Audio CD is 650 Mb (Megabyte) or roughly 0.65Gb so this should be enough for most people. A common use for Pristine Sounds is to edit waveforms for transferring to Audio CD’s

Loading, creating and saving waveforms.

To edit a waveform it must first be loaded into the workspace of Pristine Sounds. The most common file formats can be recognized, namely

- wav
- au, snd
- txt
- raw
- mp2

In Pristine Sounds, an opened waveform has a window attached to it with the name of the waveform on the title bar.

Waveform data can be distributed onto 2 temporary directories, for example on two smaller hard disks, two partitions or two computers connected an a network. Waveform data is automatically stored in the “First temporary directory”, and in the “Second temporary directory” when it cannot fit in the first directory.

> When loading a waveform, the initial data must be able to fully fit one of the directories; it cannot be split onto both. A 200Mb file cannot be split in two directories each capable of storing for example 190Mb. When applying an effect to half of such a 200Mb file, 100Mb of new storage space is needed. That data could be stored on any of the mentioned storage spaces.

Any action in PS 2000 creates an additional temporary file for the associated data. This file must fit in any of the temporary directories.

Opening a pre-recorded waveform

To open a waveform that already exist, enter the menu File->Open. Then select the file format, and open the file.

The file will be analyzed and copied into the temporary storage spaces. (Depending on the file format, it is not necessary that the storage space is equal to the initial file size. This is because PS 2000 has an own internal data format.) If the waveform has been opened in PS 2000 before, and it is of a *.wav format, it can be “Quick loaded", which is faster, and will not need a first temporary file.

Recording a new waveform

An additional way to load a new waveform into the workspace of PS 2000 is to accept data from a soundcard instead from a file. To do this you could either enter the menu 'File -> Record new waveform' or pressing the record button on the tool bar beneath the menu. You will then operate a recording tool for getting the data from your soundcard.

Saving a waveform

To be able to use the waveform in any other program than PS 2000 it must be *saved*, converted into any of the file formats described below. This must be done when a waveform is to be burned to CD-R/RW, edited in another editing program or the computer is to be turned off.

There are a couple of ways to save a waveform, namely in whole or in pieces. To simply save the whole waveform as it is, enter the menu 'File -> Save' and it will be saved with the same name and file format is it was loaded. If you wish to save with another name, please use the 'File -> Save as...' menu choice.

To save only a selected part of the waveform, you can make a selection of that part and enter the menu ‘File -> Export selection as...’ or press the right mouse button and choose to ‘Export selection as...’ This will only save in *.wav file format. (You can not export to any other format.)

A third way of saving is to “split and save in Audio CD format”. This function will split your waveform and save it in many cut-up pieces. The pieces are cut at locations that are extra suitable when creating an Audio CD because the Audio CD standard has some not-so-obvious restrictions.

Working with waveforms

Viewing waveforms

When you have loaded or recorded a waveform, it is opened in a waveform window. There you can view it in a variety of ways.

The waveform can be displayed in two major ways. Common to those are that the time line goes from left to right, and that the left channel is viewed over the right channel in a stereo waveform.

Normal view

The waveform will be displayed in the “normal view” when just loaded or recorded. The full waveform will be seen. As stated previously, the time line goes from left to right, and a stereo waveform is divided into a left and right channel.

A sound wave is really only variations of the sound pressure in the air. The *normal* waveform view draws the air pressure as it varies in time (from left to right = from the beginning to the end). To see in detail how the air pressure varies, zoom in a little bit at a time until you cannot zoom in any more. This can be done with the ‘Arrow up’ key. To zoom out, please use the ‘Arrow down’ key. When you cannot zoom in anymore, every pixel on the screen represent one sample of the waveform. An Audio CD which has 44.100 samples per second has exactly 44.100 variations on the waveform per second. If you zoom in maximally you can probably see about 1000 samples in the window which would be approximately 1/44 of a second.

If you instead zoom out maximally, it is probably not possible to see every variation of the waveform, and if the zoom out factor is greater than 200-300 samples per pixel, one can instead discern the approximate *volume* of the individual parts of the audio.

To zoom in, use ‘Arrow up’ key
To zoom out, use ‘Arrow down’ key

Frequency view

A way to discern the approximate frequencies that build up the sound is to use the *frequency* view. To make up for the fact that the frequency view can be rather incomprehensible, its at least quite beautiful! As stated above, the time line goes from left to right. In the frequency view, the low frequencies are found at the bottom of the screen, and the high frequencies higher up. (Remember that the window is divided into two parts if you are watching a stereo waveform.)

A part in the picture that is dark, has a low component of that frequency at that time. A bright part of the picture has a strong component of that frequency.

As you probably notice, the picture is brightest in the lower parts, meaning that the lower frequencies (the bass) is stronger than the higher frequencies (the treble). When air is vibrating quicker, we are saying that the frequency is higher, and when the air pressure is varying slower, we are saying that the frequency is lower. As we can see in the frequency view, the bass is more prominent than the treble, and that is simply because it takes more energy to make something vibrate faster.

Using selections

Such an easy section! Just click on the waveform and drag! This is true, but whereas there are other more accurate ways described down below.

A selection can be used for many things.

It can be used for playing

If you wish to play only a small part of the waveform in your speakers, you can make a selection and press the ‘Play selection’ button on the tool bar.

A selection can be used for controlling the viewing

To view only a small part of the waveform in the wave-form window, you can make a selection of that part and enter the menu ‘View -> Zoom in selected’

The selection is important when applying effects

If you wish to apply an effect to only a small part of the waveform, you make a selection of that part and enter the effect in the appropriate menu.

When you press the ‘Ok’ button in the functions dialog window, you will automatically be shown an additional window, the “Effect mixer”. Here you can choose to apply the effect to either the waveform in full, or only your selected part. You can also choose to use the effect on only one of the channels (if in stereo), and the fade in & out time (the time it takes to fade in the effect).

Playing waveforms

As with any simple thing, it can be done in a million strange and complicated ways.

“Playing” the waveform means that the data in the waveform window is sent to your soundcard and played through your speakers. The waveform may in some cases be modified on the fly for the best possible listening experience.

Real-time parameters toolbox

The soundcard will be set to the sample frequency of the waveform, but it can be played faster or slower if you so wish. A reason to play the waveform faster could be to “fast forward” the playing for a couple of second until you reach the desired point. A reason to play the waveform slower could be to pinpoint the exact moment when something to happens, or to investigate in depth a certain moment of time. To play the sound faster or slower, please use the “Real-time parameters toolbox”. If you have closed it down, it can be found in the menu at ‘Special -> View real-time parameters toolbox’.

In the Real-time parameters toolbox you can find other parameters for controlling the playback. The ‘Volume’ is directly linked to the soundcard and is the same parameter usually found in the “Volume Control” at the system tray. The ‘Gain’ is a software amplification of the sound to make it easier for you to investigate very silent sounds.

Between the waveform and the soundcard

Pristine Sounds has something called “Sound System Profiles” that can be inserted between the waveform and the soundcard. This is a real-time filtering algorithm that tries to adjust for the fact that the total audio peripheral system may not be perfect in all kinds of computers, stereos and speakers.

Creating a Sound System Profile

In Pristine Sounds, a Sound System Profile, consist of 32 calibration points along the frequency axis, ranging from 40Hz to 10kHz. The calibration points have been strategically placed, according to the perception of the ear.

The calibration is subjective; it depends on who makes the profile. This is because the ears of a person may just as well as the speakers or amplifiers be less than perfect. Different professional users may produce different profiles, that according the their experience is perfect.

The principle of creating a profile is to decide whether or not the 32 test tones are louder or weaker than the reference tone. The reference tone is 1kHz and is played along with the test tones. Depending on the response of the audio equipment, the sounding tones may not be equally loud. The resulting volume of the test tones will be used for creating a filter that precisely adjust for these variations.

Initially, the test tones are set to be very low (24 decibels weaker than reference tone). Going through the 32 tones, you are supposed to set the tones to be “as loud as” the reference tone (1 kHz). It is the "as loud as" definition that is subjective. It is very hard to compare the volume of a very low frequency-, and high frequency tone. Especially the low frequencies less than 200Hz are hard to estimate. What you will have to think about is: *Listening at the two pulsating tones - which one is the loudest - which one is the weakest?* It is normal not to be able to hear the difference of a decibel or two.

Press Begin. The first test tone will pulsate together with the reference tone. Use slider to make the test tone "as loud" as the ref. tone. When you are content, press "Next tone" to calibrate the other test tones. When you are satisfied with all 32 points, press “Continue...” to continue with the wizard. Enter a description for your profile and save it. When you have saved it, you will have to enable it in Pristine Sounds.

What can you expect when calibrating your equipment? If your sound system was perfect (as well as your subjective decisions...) you would get 32 tones calibrated to 0 decibel adjustments, all lying on the 0 dB line. But this never happens. Usually, a pair of headphones may have an extra strong 2-5kHz response so that you will lower the calibration at that band.

If you are unsure and cannot decide if the test tone is louder or weaker than the ref. tone, please set the calibration closer to the 0dB line.

Using a Sound System Profile

To enable an existing Sound system Profile, you can enter the menu ‘File -> Sound System Profiler...’. Browse to the Profile and enable it by checking the ‘Enable Sound System Profiles’. The profile can also be enabled or disabled in real-time by checking this option in the real-time parameters toolbox.

Applying effects & functions

There are a multitude of ways to modify the waveform.

Some functions may be applied to parts or to the whole of the waveform. Other functions cannot be applied to only parts of the waveform. Such functions would include changing the global properties like the sample rate, number of channels etc....

Changing the global properties of a waveform

Some functions affect the global properties of the waveform, but do not affect the actual sample data. Such functions include

- Changing the sample rate
  A well defined number of samples has to be transferred to the soundcard every second. Normal Audio CD waveforms have 44.100 samples per second. Changing the number of samples will affect the playback speed; if the number of samples per second is reduced, the playback will be slower and the audio will descend into the lower frequencies; if the sample rate is risen, the playback will be faster and the audio content will transcend into the higher frequencies.

  (When you wish to change the sample rate without changing the speed of pitch, the waveform has to be “re-sampled”, and this will affect every single sample in the waveform.)
- Changing the number of channels
  A waveform in Pristine Sounds has either 1 or 2 channels. To change a mono sound into a stereo sound, the waveform data will be doubled, so that the new “left” and “right” channel both contain the old “mono” channel. This will not affect the audible stereo effect of the sound; to broaden the apparent stereo experience, special functions has to be applied.
  To change a stereo waveform into mono, the 2 left & right channels are merged into one mono channel. This is simplest done by taking the mean out of the two channels. However, a special function exist in

the Vinyl Restoration Tool (VRT) to make this function more refined. [See ref]

It is not possible to have different sample rates in different parts of the waveform. Neither can one waveform has 1 channel at one place and 2 channels at another place.

Applying functions & effects to the waveform data

An “Effect” in Pristine Sounds is defined as a modification of the existing waveform data, where no samples are added or removed. (Other functionality may change the number of samples, for example adding or removing silence.) *All effects also fall under the category “function”, which simply is a way to modify anything in a waveform*.

Examples of effects are (among many others)

- Changing the volume
- Adding echoes and reverberation
- Filtering
- Removing noise & vinyl scratches

Any of these effects can be applied to the whole waveform or a part of it.

To apply an effect to only a part of a waveform, make a selection before entering the effects dialog window, and after pressing ‘Ok’, use the Effect Mixer to choose ‘Selection’.

The common interface of all effect windows

Most effects have adjustable parameters. Excluded is for example the ‘Normalize volume’ that only has the task to rise the volume in such a way that it do not clip any samples. The ‘Remove DC-offset’ do not have adjustable parameters either, as it simply apply a 5Hz high pass filter, thereby removing any baseline offset of the waveform.

All effect windows have an ‘Ok’ button and a ‘cancel button. After pressing ‘Ok’ yet another window, the ‘Effect Mixer’ will be displayed. In the Effect mixer you define how to apply the effect (fade in & out times, channels to apply to etc.). The Cancel button will leave the waveform untouched in its current state.

Most effects can be previewed. This mean that you can take a copy of a small segment of the waveform and apply the parameters to that part, without affecting the actual waveform data. This can be done any number of times and is used when trying out which parameters may be the optimal. [See ref] In Pristine Sounds 2000 Pro, most effects can also be previewed in real-time, meaning that a part of the waveform is played in the background at the same time you change the parameters, therefor hearing immediately what the outcome will be.

Real-time previewing is not available in Pristine Sounds 2000 Light

The help file for the effect can also be accessed by pressing the ‘Help’ button.

Many (but not all) effects also have ‘Load’ and ‘Save’ buttons. There are two reasons for that. If you have found a particularly good set of parameters, you can save them for later usage. A second reason for saving parameters is for the usage in “batch mode”. In batch mode (or “Automatic treatments”) you can apply almost any effect in any order, totally in groups of 8 effects. This can be done automatically without user assistance, and can be applied to up to 128 waveforms. When defining which effects to “batch execute” you also need to define the actual parameters saved by you.

Batch processing is not available in Pristine Sounds 2000 Light

The Effect Mixer

When you have pressed the ‘Ok’ button in the effect window, you will be asked to input some information about *how* to apply the effect. Usually, the effect has already set those parameters for you. The parameters are:

- Fade in & out time: This will set the fade in & out time for your effect. *(This property is purely additive; if the effect is a "pitch bend" this will fade between the original sound and the "pitch bend" sound - it will not slide between notes.)*
- Amount of effect: This will set the total amount of "effect signal and original signal" that will be mixed together. (If set to 80% the processes signal will account for 80% of the result, and the original signal 20%.)
- Selection / full piece: This will define *where* to apply the effect. If you have a selection, you may apply to either the full waveform or only the selection.
- Left / Right channel: If you have a stereo waveform, you may apply to either left or right channel or both.
- Residual output: This is an option for testing purposes only. The resulting output is the difference between the original and the effect.

  Example:
  1) When removing noise, the residual output is the noise itself.
  2) Then applying reverb, the output will be the reverb part.
  3) When using a filter, the output will be the added components only.

- Soft clipping. The internal format of the sample processing in Pristine Sounds is 24 bits. This gives a big headroom for converting the samples back to the CD-standard of 16 bits when the processing is done. It implies that the samples can exceed 16 bits internally without actually clipping. The Soft clipping options applies an algorithm on those extra 8 bits so the conversion goes extra smoothly. Note however that older DirectX / ActiveMovie plugins work in 16 bit mode only and take care of the clipping themselves.

DirectX plugins

Pristine Sounds support a standard of effects called “DirectX” plugins. (Also called DirectShow or DirectMedia. The last version of the standard was called Active Movie. So much for standardization.)

A DirectX plugin can be installed long after Pristine Sounds was installed. Every time PS 2000 is starting, it searches the system for new DirectX plugin effects. If any can be found they will be listed in the Add-ons menu. Alien Connections only instantiate those effects into Pristine Sounds. The responsibility for the accuracy and operability of those effects lie on the manufacturer them selves. In the same ‘Add-on’ menu all plugins using the native Pristine Sounds plugin standard are listed.

When selecting a DirectX plugin, it will be opened in a DirectX effect window. It is quite similar to the ordinary effect windows, as they have the same ‘Preview’, ‘Load’ and ‘Save’ buttons. All other buttons however are local to the DirectX plugin itself and are not controlled by Pristine Sounds.

Sometimes, some DirectX plugins are listed, but grayed out, in the ‘Add-ons’ menu. This is because the effects cannot process the currently selected waveform. It is probably because the effect cannot process a waveform of that particular number of channels. *(A stereo reverb cannot process a mono waveform and vice versa.)*

The preview dialog window

Almost every effect can be previewed.

You can test your settings so they appear as you intended. This automatically takes a couple of seconds of your waveform and applies the effect to it. The actual waveform will be untouched.

This is easier than applying the effect to some part of the waveform and making an UNDO if you are not satisfied.

Real-time previewing is not available in Pristine Sounds 2000 Light

Under the "Play original"-, and "play effect" buttons there are some statistics about the waveform: Maximum & minimum sample value, RMS. and the number of channels.

Play buttons: Press to hear either the original or the processed version of your waveform. The "Play original" and "Play effect" should be obvious, but the "Play combination" deserves some explanation: This enables true A/B listening. Define what will be played in each channel: the left/right original sound or the left/right effect sound. If you choose left original as the output on the left channel you will hear the originals left channel in the left speaker. Choose left effect in the right speaker to hear the effects left channel. For example; if the effect in question is noise suppression, you will hear the left channel in the middle and some noise in the right speaker, and no (preferably:-) noise in the left speaker.

Stop buttons: Press to stop waveform playing.

2 / 4 / 8 seconds: The amount of time of the waveform that the preview will use.

Position slider: Set the actual point of testing. *(This point will be remembered the next time you enter the preview window using the same effect & waveform.)*

Waveform / Spectral view: Use the waveform view to see the waveform amplitude, or use the spectral view to examine the signals frequency components.

Intensity: For use with spectral view, to brighten or dim the view.

Graphical windows: A quick snapshot of the original and the processed waveform.

Volume...: Displays the "real-time parameters" toolbox.

Residual output: This is an option for testing purposes only. The resulting output is the difference between the original and the effect.

Example:

1) When removing noise, the residual output is the noise itself.

2) Then applying reverb, the output will be the reverb part.

3) When using a filter, the output will be the added components only.

Real-time... By pressing this button, the preview dialog window will fold down into a small toolbox beside your current effect window. You can now use the controls of the effect window and hear the result playing at the same time. By pressing 'Stop!' you exit the previewing mode.

Real-time preview toolbox

The real-time preview toolbox has the following parameters:

Bypass processing: Use this function to switch between the actual processed waveform and the original. This can be used for exactly pinpointing the performed work.

Left channel only: Use this function to process only one channel. This will make processing twice as fast when previewing stereo waveforms. *Some effects can not use this trick.*

Slow playing: When the CPU's processing power is not enough, the playback is automatically slowed down. Using a really low playback rate from the start ensures that the rate will fluctuate less. A varying playback speed can be annoying, and this option makes playback constantly low.

More on special features

Undoing actions

Alas, not everything we do is right. Sometimes you will need to undo your actions. Pristine Sounds can remember everything you do, so that you can select an older state of the waveform. To undo an action, please enter ‘Undo action’ from the ‘Edit’ menu. To reselect a newer state, enter the ‘Redo action’ menu.

If you wish to navigate more exactly among the history of all states, please enter ‘Action navigator’. You will then be able to select among the states in a graphical tree.

To save hard disk space, Pristine Sounds is set to delete data that belongs states that is older than 25 actions ago. You can set this number in ‘Preferences’.

Note! By setting the ‘Minimum number of states’ you can ensure that this number of states is guaranteed to be available. It is quite possible that a larger number of states also are saved than what you ask for. The reason for this that some states are directly dependant of states older than the set value. In a system of non-linear editing, a “state” can consist of data from many other states.

Recording a waveform

Pristine Sounds offer some functionality for recording through any Windows compatible soundcard. Although it is possible to do the sampling with Windows built-in tools (like soundrec.exe et al) or programs that came with your soundcard, it is recommended to use Pristine Sounds recording tool as it has some extra features. The following descriptions assumes that you have opened up the recording tool, either by pressing on the recording button on the tool bar or the menu choice 'Edit -> Record new waveform...'

Select your recording device

If you have more than one soundcard (or a soundcard with more than one input devices) you can choose which one to use. Press 'Set properties' to select the input device, sampling frequency and number of channels. Why can't you choose the number of bits per sample? Most soundcards can sample with 8 and 16 bit accuracy, but the ability to use 8 bit samples do not serve any purpose today. The quality is terrible. In future versions of Pristine Sounds there might be support for soundcards that can sample with 18, 20 or 24 bit samples as well.

The soundcard frequency analyzer

As well as common VU meters, you can also analyze the actual frequency content in the incoming signal. This can be useful when tweaking the settings before recording. As opposed to the VU meters the frequency analyzer will not analyze the signal after background processing, only data *directly from the soundcard*.

The soundcard frequency analyzer displays in real-time the content of the incoming signal. On one axis you have the frequency, and on the other the strength of the frequency.

Hard disk space

When you record a new waveform, the data will automatically be stored in the *Primary temporary directory.* (You must make sure that this space can hold the length of your record.) If the empty space is 5Mb or less, all recording will stop! (5Mb is required to ensure that Pristine Sounds will work correctly.) When the file size of the recorded file is 2Gb or more, the recording will stop as well. The reason for this maximum file size is historical and built into Windows.

The VU meters

To ensure that the incoming signal is not to weak or not to strong, you can monitor the levels with the VU meters. You can also monitor the levels before or after background processing. (If you apply a strong filter, the volume may change drastically.)

The levels will be sampled faster before recording than after you press ‘Start!’.

If the incoming signal is being clipped, a warning will be asserted and after you pressed ‘Start!’ this warning will be persistent (it will not go away).

Background processing

To save time, some of the built in effects in Pristine Sounds can be applied at the very moment of recording. The available effects use the same settings as they do inside of the program. You will have to enter this effect / function and design a setting that you think is suitable for the sound you are going to record.

When recording a vinyl album, suggested background processes are

1) The VRT

2) Image noise suppression.

(DC offset removal is *always* suggested.)

You can have up to 2 background processes running at the same time. It is not guaranteed that the CPU can keep up with the incoming data. If the mean CPU utilization is noticeable less than 100% it is quite safe to assume that no signal will ever be lost. If the CPU utilization is between 98-100% you should remove some background process or apply a setting that could be simpler.

It is not possible to have one effect assigned to both background processes. (Although it might be possible to make a duplicate copy of the *.Plugin file in the plugin directory.)

Program switches

To start Pristine Sounds, you can either double click on the icon on the desktop or in the Start menu. You can also start the program by executing it with the Run command on the Start menu or in a command prompt.

When Pristine Sounds is started via its program name, extra switches are available. The switches can be used for sending commands to the program.

Please note that the program “Pristine Sounds 2000.exe” is just a program loader; this is not PS 2000 itself. If PS 2000 is not running, the loader will start it, and then send the command line to it via the SendMessage API. The exit code of the loader program is the return code of the message sent to PS 2000.

Exit codes, aka return values associated with the program switches can be found in the file “remote_control.h” file in the \Extras directory.

Because only one instance of PS 2000 can run at a time, it is possible to send more than one command to the program. Those commands will be sent to the same instance of the program. Please use the habit of starting up the program without program switches.

A call with a program switch will return when the operation has been completed if not otherwise stated.

These switches are defined for starting / sending commands to PS 2000:

Filename

This will load the waveform into Pristine Sounds.

/RECORD

This will start the recording of a new waveform in PS 2000. The currently selected parameters will be used. The call will return when recording has started.

/PLAY_ALL

This will play the current waveform from beginning to end. The call will return when playing has started.

/STOP

This will stop all playback or recording if such was running.

/BATCHPROCESSING_RESET

This will prepare automated batch processing. All scheduling of files and treatments will be reset. Saving of files will be directed to the original directory (files will be overwritten). Playing of processed waveforms is disabled. Files will be closed when done. This *must* be done before adding files or treatments.

/BATCHPROCESSING_LOAD_TREATMENT Filename

The treatment will be loaded and scheduled for batch-processing. Only one treatment can be active at a time.

/BATCHPROCESSING_ADD_WAVEFORM Filename

The file will be scheduled for batch processing. Up to 128 files can be batch processed at a time. The file will be loaded when batch processing is started.

/BATCHPROCESSING_RUN

The treatment will be executed. The call will return when all files have been processed.

/BATCHPROCESSING_ERRORCODE

This will return the error code from the processing.

/COMPUTER_UNMANNED boolean

When the boolean is **TRUE** the computer will be considered to be unmanned. All user interaction in the form of popup windows will be suppressed. Please use this switch with cation. You will never know when something has gone wrong!

/SCAN_NOISE filename

Assuming that a waveform is opened, it will be scanned for noise, and a noise image is created with the given filname. The following defaults are used:

* >5 kHz, 25%
* 1-5 kHz, 30%
* <1 kHz, 35%
* 50 Hz, 100%
* No carving
* Release time, 90
* Overwork protection, 5

/QUIT option

This will quit the program. When the option is **SAVE** all opened files will be saved, otherwise they are closed without saving.

The following example could be executed either from the "Run" start menu, or from the command prompt or a batch file:

For recording, please use the following procedure. It can be useful, when starting a recording on an unmanned computer.

```
"Pristine Sounds 2000.exe" /RECORD
"Pristine Sounds 2000.exe" /STOP
"Pristine Sounds 2000.exe" /PLAY_ALL
"Pristine Sounds 2000.exe" /STOP
```

If PS 2000 was not started, it would start at the first line. It would then start recording with the currently selected parameters. When the /STOP line is reached the recording will stop. Playback will then be played from the start and will be stopped either when the /STOP command is given or when the end of the waveform is reached.

This is the recommended procedure for batch processing:

“Pristine Sounds 2000.exe” /BATCHPROCESSING_RESET
“Pristine Sounds 2000.exe” /BATCHPROCESSING_ADD_WAVEFORM c:\temp\test.wav
“Pristine Sounds 2000.exe” /BATCHPROCESSING_ADD_WAVEFORM c:\temp\test2.wav
“Pristine Sounds 2000.exe” /BATCHPROCESSING_LOAD_TREATMENT c:\temp\test.treatment
“Pristine Sounds 2000.exe” /BATCHPROCESSING_RUN

The procedure starts with a reset. Then the waveforms and the treatment are loaded, where after the batch processing is started. The /BATCHPROCESSING_ERRORCODE could optionally be inserted after execution to read out the result of the operation. If another batch is to be processed, you must restart with a reset.

Understanding quick loading of files

Pristine Sounds 2000 can use “quick loading” of a waveform if:

* It has been previously loaded into PS 2000.
* It has not been altered by another program since.
* It is a (16 bit PCM) *.wav

If this is true, it will be possible for PS 2000 to load the file only “virtually” instead of transferring the file into a copy as a temporary file. This speeds up loading tremendously if the waveform is big.

In fact, the original file it self will be used as a temporary file. This mean that you can only have one copy of the waveform opened at a time.

If the *.wav file has never been loaded into PS 2000, or if the file has been altered outside the program, the quick-load feature will read in only the minimum amount of information that is needed to display it as a waveform in the window. When this is done, you can work with the waveform as usual, although PS 2000 will use idle time to read in the remaining information. (This is mentioned as “Refreshing peak file”.) Using this technique, a 650Mb file can be loaded within 15 to 20 seconds, not using any temporary space.

Using “Skins”

PS 2000 has the ability to use “skins”. This is what commonly is known as a bitmaped surface of the windows.

Naturally, this only enhance the visual appearance; it will not speed up the processing...

The bitmaps you can use for skin effects can be found in the ‘\Skins’ directory under the installation directory.

The \Skins\Windows directory contain bitmaps for the surface of all windows.

The \Skins\Backgrounds directory contain bitmaps for the background of PS 2000.

The bitmaps in the \Skins\Windows directory will be adjusted to the current color theme you might have. This will not happen to the background bitmaps.

If you wish to use your own skins, the format of the bit-maps is:

* A Windows *.bmp bitmap
* 24 bit color
* A maximum of 16384 pixels, in any dimension (for example 128x128 or 256x64) It can be smaller than this, but not larger.

The bitmap will be tiled so you should make sure it is periodical.