weiss

GAMBIT
DNA1
OPERATING MANUAL

Block diagram:

AES/EBU CH1/CH2 → L/R to M/S Encoder → Declicker → Smooth → Denoiser

M/S to L/R Decoder → K-Stereo → Gain Control → POW-r → AES/EBU CH1/CH2

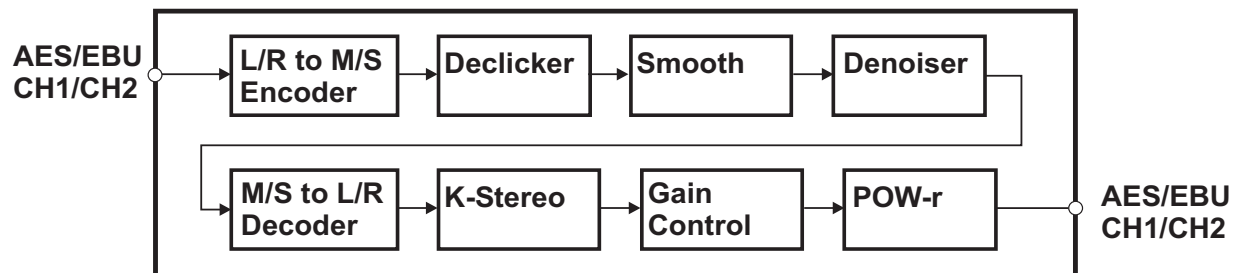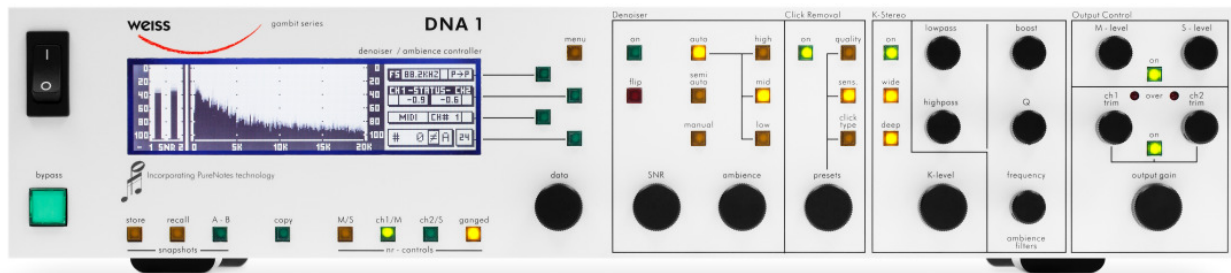| Software Version:	OS: 3.1, DSP: 3.0
Release Date:	November, 2007

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CONTENTS

FRONT PANEL SECTIONS

Fig. 1 Front Panel and Display Sections

Front Panel Sections

The front panel of the DNA1 offers several control and display features. These can be grouped according to Fig. 1 Front Panel and Display Sections:

❶ - Snapshots (p. 11)

❷ - Noise Reduction Controls (p. 23)

❸ - Menu and soft keys (p. 10)

❹ - Denoiser (p. 13)

❺ - Declicker (p. 13)

❻ - K-Stereo (p. 25)

❼ - Output Control (p. 26)

❽ - SNR Meter and Quality Meter (p. 19 and p. 17)

❾ - Real Time Spectrum Analyzer with Noise Threshold Display (p. 19)

❿ - Status Display (p. 10)

This manual will explain how to operate the DNA1 according to these sections.

The Bypass key is discussed separately in chapter Bypass (p. 27). The "data" encoder appears throughout the rest of this manual and is therefore not explained separately.

INTRODUCTION

Congratulations on purchasing the Weiss Gambit DNA1 Denoiser / Ambience Controller!

In cooperation with the companies Pure Notes (Denoiser) (www.purenotes.com) and Digital Domain (K-Stereo) (www.digido.com), Weiss Engineering has developed a state-of-the-art Denoiser / Declicker and Ambience Enhancement unit, the Weiss Gambit Series DNA1.

Denoiser / Declicker

The DNA1 uses a novel approach to noise reduction that can remove a large range of noise types without affecting the musical characteristics, stereo imaging or the ambiance of the original. There are two basic functional parts:

1. Declicker
2. Denoiser

If the Declicker is enabled, the system automatically detects and removes impulsive noises such as pops (clicks) or static (crackles) from vinyl recordings, as well as very fine single sample ticks. Declicker is followed by a Denoiser that reduces slowly varying (or constant) noises such as tape hiss, traffic and wind rumble, air conditioning etc. To achieve this, time and frequency content is analyzed to optimally isolate and remove the offending noise. The process can be fully automated, or carefully hand-tailored with a variety of parameters.

While the DNA1 represents the best available technology for noise reduction, incorrect or harsh
settings can force the system to overdo the reduction and therefore affect the quality of the music. There are theoretical limits to the information content of a noisy signal, and in severe cases there may be very little signal information available. The DNA1 can be used to enhance the signal and improve the SNR as much as possible.

K-Stereo

For the first time, the mastering engineer can enhance, expand, and equalize the ambience in a recording. The nicest thing about K-Stereo is that it works unobtrusively and naturally. Do you have a "big" rock and roll recording that's too "small"? You can fix it, avoiding a costly remix. You will find K-Stereo an extremely natural-sounding stereophonic enhancement technique during post-production, mastering or even mixing. K-Stereo gives you control over your reverb returns or ambience mikes *after* the program has been mixed, and it enhances the shape, spread, and depth of that reverb.

K-Stereo is unlike any of the so-called "re-imaging" processors you have heard. While other boxes seem to have a sound of their own … this processor fills in what's been missing. K-Stereo initially appears subtle. Don't expect an out of body experience! Instead–you'll hear a natural, diffuse ambient field derived from the ambience in the source recording. K-Stereo enhances a recording's clarity, spatiality, depth and soundstage without creating phasing or comb-filtering effects, without matrixing or altering mid/side ratio, without changing the "mix", without fancy steering, and with no effect on the tonal balance of the direct sound.

… And More!

In addition, the DNA1 has separate knobs for mid/side, left/right and overall volume control. Also included is the set of acclaimed POW-r word length reduction algorithms for highest quality 24 bit, 20 bit or 16 bit digital output.

The large high-contrast LCD shows a real-time spectrum analyzer, SNR/SNR Improvement meters, declicker quality meters as well as signal status parameters, over and peak holds.

The DNA1 is fully remote controllable, either with snapshot recalls or with direct parameter access, over MIDI, RS-232 or RS-422 remote control ports.

OPERATION

> **The following explanations assume that the DNA1 is in power-up mode, i.e. no menu is active. If a menu is active, press the "menu" key repeatedly until the status display appears.**

Status Display
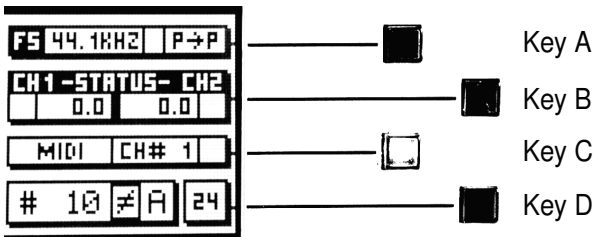
Fig. 2: Status Display

The status display (display section **10**) consists of four groups displaying audio signal properties and information about the current state of the DNA1. The groups are situated adjacent to the soft keys to the right of the LCD (Fig. 2: Status Display).

Channel Status Group

Situated adjacent to key A, displays the following information:

- sampling frequency in kHz: "44.1", "48", "88.2" or "96"

- Emphasis: "E" appears if emphasis bit is set (empty if not used)

- input → output channel status data format: "C" for consumer and "P" for professional

To change the output channel status data format, press key A and select format type or loop through.

Status Group

Situated adjacent to key B, this group contains a peak-hold and an over-hold for each channel. To reset these values press key B twice.

Peak-Hold

The numeric peak value is measured relative to 0dBFS, where 0dB equals digital full scale.

Use this value to set the optimal gain for a specific session. Reset it for every session.

Over-Hold

Whenever an over is encountered, the boxes to the side of the peak-hold values are marked with a dot.

The functions of the over-LEDs and the over-hold both depend on the NUMBER OF CONSECUTIVE OVER-SAMPLES setting. To adjust this setting, press key B once. Use the "gain/data" encoder to change the value, then press "menu" to return to the status display.

This setting defines the number of consecutive over-samples needed for an over-indication. To indicate every over-sample, enter 1. If another value is entered, you are still able to detect a single over-sample with the peak-hold function.

> **Over-samples are defined as exceeding the 24 bit integer range. A full-scale signal (0 dBFS) will not be treated as over. A signal that triggered an over in a previous processor in the audio chain will therefore not trigger an over in the DNA1 (unless boosted).**

Remote Group

Situated adjacent to key C, displays remote control protocol ("MIDI", "RS 232" or "RS 422"), channel ("1"–"16") and status (on/ off, displayed with check box).

For remote control set-up see chapter Remote Control on p. 27.

Snapshot Display

This is the bottom line in the status display. Refer here for information on current workspace and snapshot. There are two workspaces ("A" and "B") for quick comparison of two settings. A workspace can be stored to a snapshot for later use.

The number of the last recalled snapshot will be displayed. If you change any value after the recall, the "=" will change to "≠" to symbolize that the workspace is not equal to the displayed snapshot anymore.

For snapshot handling see chapter Snapshots below on this page.

Dither Icon

The dither icon is situated right next to key D (see Fig. 2). If dither is off, the icon contains the number "24", the word length of the output signal in bits. Turning dither on will produce arrows above and below the new word length number ("16" or "20"). Additionally, if auto-blacking is enabled, the bottom right corner of the icon is inverted.

For details on dither refer to chapter Dither on p. 27.

Snapshots

See Fig. 1 Front Panel and Display Sections, ❶.

Snapshots are copies of the DNA1 parameter settings. This allows recalling complete set-ups including all filter parameters and controls settings.

Snapshots can be recalled from the front panel or by remote control (e.g. MIDI Program Change) for automation.

A-B Comparison

You can compare two settings by using the two workspaces "A" and "B". To toggle between the two press the "A-B" key. Restore a snapshot or adjust the workspace, then switch to the other to compare.

To copy workspace "A" to "B" (or vice versa), press the "copy" key and select an option (Fig. 3: "copy" menu).
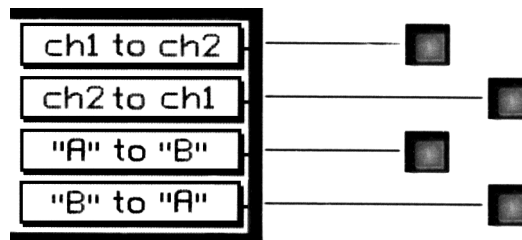
Fig. 3: "copy" menu

The active workspace is shown in the status display.

Recall

To recall a snapshot press the "recall" key. Notice that the snapshot will be copied into the active workspace. If you do not want to lose these settings, switch to the other workspace by pressing the "A-B" key.

Using the "gain/data" encoder enter the number of the snapshot you want to recall.

Press the "recall" key to return to the status display.

"Recall # 0"

Snapshot 0 is used as a resetting means. To reset the current workspace, press the "recall" key then the" Recall # 0" key.

Store

To store the current workspace to a snapshot, press the "store" key. With the "gain/data" encoder enter the snapshot number where you want to store the workspace, then press key D. Snapshots can be prevented from accidental erasure by marking the "Safe" option.

A setting can be stored to multiple snapshots. Press "To Range" in the "store" menu.
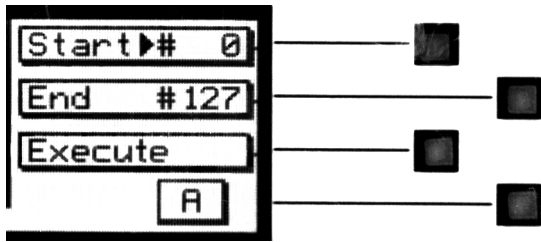
Fig. 4: Store To Range Menu

Press the adjacent keys to select "Start" and "End" snapshot number, use the "gain/ data" encoder to enter a value (Fig. 4: Store To Range Menu). Pressing "Execute" will store the setting from the displayed workspace ("A" or "B") to all snapshots between the Start and End value. A "Safe" snapshot will not be overwritten.

Backup

All 128 snapshots can be transferred to another non-volatile memory area for later reference. There is enough memory for two complete snapshot sets to be backed up.

Press the "menu" key then choose the "System" and then the "Backup" option to arrive at the backup display (Fig. 5; Backup Display).
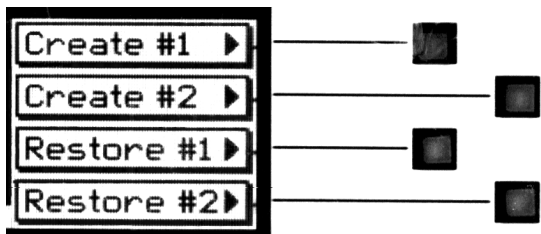
Fig. 5; Backup Display

Select "Create #1" to save the current snapshots to bank 1. Select "Restore #1" to overwrite the current snapshots with bank 1 (ditto for bank 2).

If the "Restore" option is not available, then no backups are stored in memory.

> **A backup will overwrite "safe" snapshots.**

Reset

To reinstall the factory snapshots you must press the "menu" key then choose the "System" and then the "Reset all" option. This will reset the current snapshots. The snapshots that have been backed up are preserved.

Dumping

The settings of all 128 snapshots can be dumped to a remote control port to be stored externally. Please make sure that you have connected everything correctly and that the proper remote control port is activated (see chapter Remote Control for set-up).

When dumping to a MIDI-sequencer, make sure that the sequencer is not sending or receiving data other than the snapshot dump.

> **Certain sequencers are not capable of handling the amount of data output by the DNA1 and might crash during or after the dump process.**
>
> **Always save the sequencer settings before recording a dump!**

To initiate a dump, press the "menu" key then choose the "System" and then the "Dump" option. Start the recording process on the sequencer, then press "Transmit". This releases the dump to the sequencer.

Loading

To load a snapshot dump from sequencer, simply play the dump sequence. The DNA1 will automatically switch into receive mode and display statistics about the snapshots it's receiving. If the transmission was OK and no errors occurred, the program switches back to normal mode. Upon error, you have to manually switch back to normal mode.

> **When playing back the dump sequence, always use the same speed as when the sequence was recorded.**

DNA1 Processing Chain

The Weiss Gambit DNA1 consists of a series of two-channel signal processing blocks, which is shown in Fig. 6.
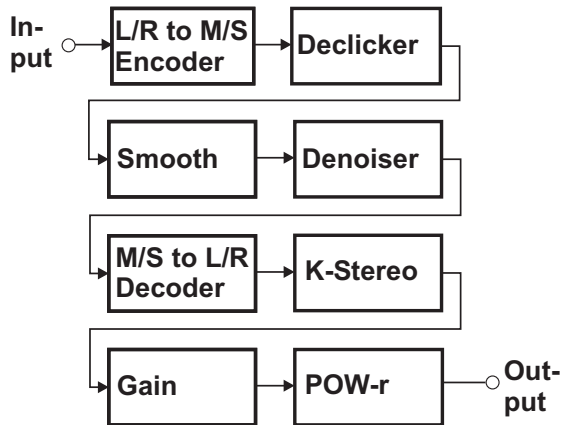
Fig. 6: DNA1 Processing Chain

The "Declick" block is controlled from the Declicker section (**❺**) in the front panel. "Smooth" can be adjusted from the "menu"   "Declicker" sub-menu. The "DeNoise" block is controlled from the Denoiser section (**❹**). Some additional parameters are accessible from the "menu"   "Denoiser" sub-menu.

The Noise Reduction Controls (❷) only affect the Denoiser and the Declicker operation.

K-Stereo controls are located in front panel section **❻**, the output gain controls in **❼**. POW-r dithering is adjusted in the Dither menu (see p. 27). Below follows a detailed discussion of each block's parameters.

The parameter for Smooth is only adjustable for both channels simultaneously. All other parameters can be adjusted for each channel separately.

Declicker

"Declick" and "Smooth" processing blocks make up the Declicker part of the DNA1. The "on" key in the Declicker section has three states: on, hard bypass and soft by-pass. If the "on" key in the Declicker section is not lit, the whole Declicker block is skipped in this case and the Declicker latency is also taken out of the signal chain (hard bypass mode). A short push on the "on" key will enable the Declicker section and the key will light solidly (on mode).

Another short push will set the Declicker section in soft bypass mode where the audio signal is not declicked but the latency remains. The soft bypass mode is indicated by a flashing "on" key.

To activate the hard bypass again press the "on" key for about 2 seconds.

Parameter And Processing Description

The declicker part of the DNA1 is controlled by nine parameters and user feedback is provided with one value. In the following we will describe the basic processing steps carried out in the declick section and how the parameters interact with the processing.

Very generally speaking, the declicking process is carried out in three steps:

1. click detection
2. click interpolation
3. smoothing process

In the click detection process the audio is analysed and clicks located. This process is controlled by six parameters (detection threshold, extension threshold, asymmetry, adaptivity, discard threshold and maximum click length).

The click interpolation step replaces the located clicks to match the surrounding undamaged audio. It is automatically optimised by the DSP software in order to guarantee best restoration quality.

For some recordings, smoothing can be used for subtle high frequency noise reduction and one-sample ticks removal. It does not have any user controlled parameters besides a simple on/off switch.

For a graphical illustration for the first two processing steps see Fig. 7.
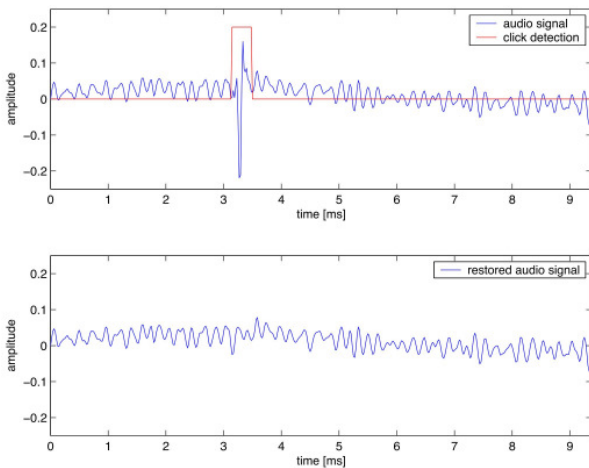
Fig. 7: Declick Process

In the following we will describe all user controlled parameters in more detail.

Detection Threshold

The detection threshold controls the main sensitivity of the click detection process. It can be set by pressing the "sens." key, selecting "DETECT. THR." (key B) and adjusting the value with the "data" encoder. With a high detection threshold only large (and hence typically only few) clicks are detected. Setting the detection threshold to a low value will detect both large and small clicks (see Fig. 8).
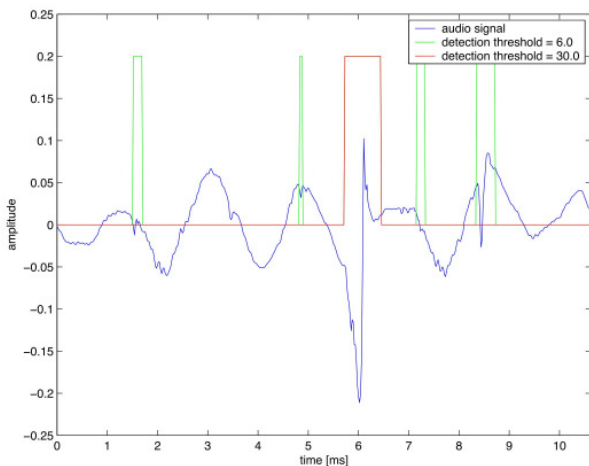
Fig. 8: Detection Threshold effect

The detection threshold should be set to the highest value which allows detection of all audible clicks. If the value is too high, not all audible clicks are detected and hence after interpolation some clicks will remain in the audio signal. If the value is too low on

the other hand, undamaged audio may be wrongly marked as clicks, which is undesirable.

From the above description it is clear that the detection threshold is a very critical parameter. Fortunately its adjustment is simple and straightforward: start with a high value and decrease it until all audible clicks are removed.

Extension Threshold

The extension threshold controls the width of the detected click region. It can be set by pressing the "sens." key, selecting "EXTENS. THR." (key C) and adjusting the value with the "data" encoder. A high extension threshold results in a narrow detection region, while a low extension threshold causes a wide detection region (see Fig. 9).
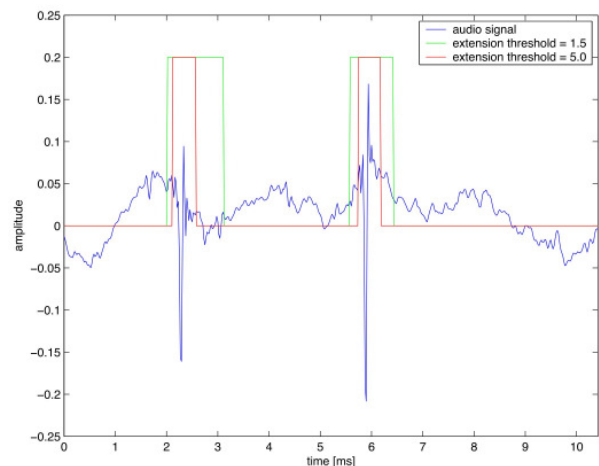
Fig. 9: Extension Threshold effect

The extension threshold should be set to the highest value that still allows clean removal of all clicks. If the value is too high, only part of the click may be detected. That way, the main transient of the click is removed in the restoration process but before and after the main transient some more subtle degradation of the audio remains. If the value is too low, undamaged audio may be included in the click region.

Extension thresholds below the optimum value will lower the interpolation quality, as larger interpolations must be calculated. Aditionally, extremely low extension thresh-

olds may result in very long click regions which will be rejected by the software.

To set the extension threshold start with a low value and increase it right below to the value where the clicks become audible. Typical values range from 2.0 for larger clicks to about 3.5 for very small clicks.

Asymmetry

The asymmetry parameter allows the user to adapt the detection process to different click types and shapes. It can be set by pressing the "click type" key, selecting "ASYMMETRY" (key D) and adjusting the value with the "data" encoder. Clicks from analogue recordings (gramophone) typically show a tail extension as a result of the impulse response of the playback system. A high asymmetry value extends the detected click region to the right in order to better fit the shape of the click. Low asymmetry values provide a more centered detection region, e.g. for clicks caused by digital sync problems. The action of this parameter can be rather subtle, but it helps to tailor the algorithm in critical restoration tasks (see Fig. 10).
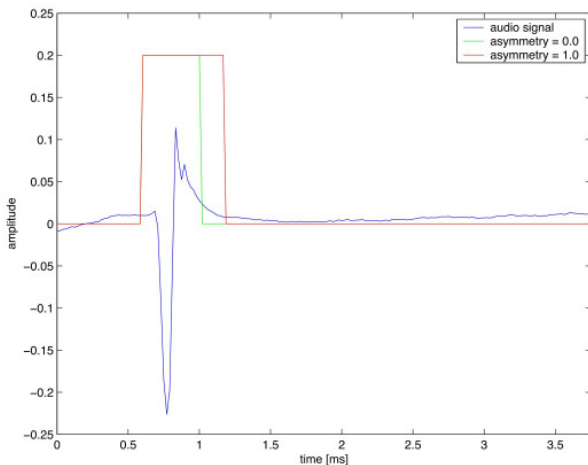
Fig. 10: Asymmetry effect

The asymmetry should be set to the value allowing for the highest extension threshold value. This will minimise the amount of undamaged audio being included in the click region. To determine an initial value, it is helpful to know the click shape. Simple visual inspection with an audio editor or alter-

natively an educated guess considering the click cause should prove helpful. Typically large clicks from analogue recording media (gramophone etc.) need higher asymmetry values (say 0.6–1.0) than small clicks (say 0.2–0.4) or clicks resulting from digital transmission problems (typically 0.0). For a further refined setting, try values higher and lower than the initial setting and see if you can increase the extension threshold.

Adaptivity

The adaptivity setting controls the speed with which the click detection algorithm adapts to level changes of the audio signal. It can be set by pressing the "click type" key, selecting "ADAPTIVITY" (key C) and adjusting the value with the "data" encoder. A high setting allows for fast adaptation, while a low values provides slower adaptation.

The need for this parameter is comprehensible when considering that large clicks may appear as a transient sound (rather than a click) to the algorithm; or the other way round, that percussive sounds (snare hit etc.) are interpreted as click. A fast adaptation (i.e. high adaptivity setting) avoids wrong detection of percussive sounds but has the danger of missing large clicks. In contrast, a slow adaptation (i.e. low adaptivity setting) greatly reduces the danger of missing clicks but in turn may wrongly detect transient sounds as clicks. To summarise: large clicks and music with slow level changes favour a low adaptivity value while small clicks and music with fast and frequent level changes are better suited for high adaptivity settings.

Usually this parameter can be chosen according to the following table:

	Slow Level Changes	Medium Level Changes	Fast Level Changes
Small Clicks	0.4	0.6	0.8

Medi-um Clicks	0.2	0.4	0.6
Large Clicks	0.0	0.2	0.4

Discard Threshold

Often the click detection process reports very small clicks (frequently only very few samples long) which are inaudible. It is desirable to not interpolate these clicks.

As the name implies, the discard threshold controls the process of discarding inaudible clicks. It can be set by pressing the "sens." key, selecting "DISCARD THR." (key D) and adjusting the value with the "data" encoder. A low discard threshold discards only the smallest and most inaudible clicks while a high discard threshold may in fact discard clicks which are audible. If the discard threshold is set to zero, no clicks will be discarded (see Fig. 11). To set the discard threshold it is best to start with a low (preferably zero) value and increase it until some clicks become audible, then reduce the value a tad.
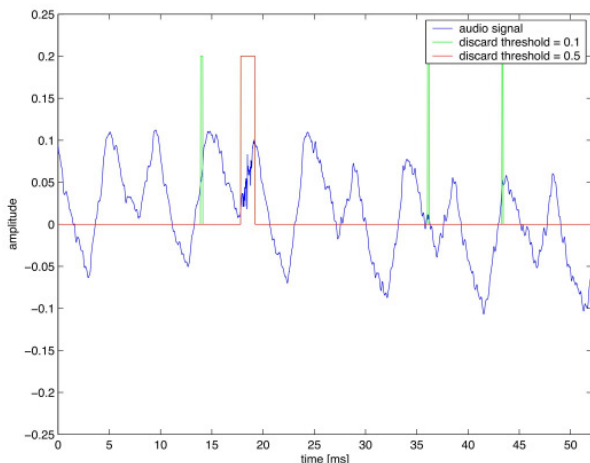
Fig. 11: Discard Threshold effect

It is important to realise that the discard threshold may interact with the detection threshold. If you lower the detection threshold and the clicks which are newly detected are small, it might happen that the discard process simply discards these clicks, i.e. there will be no audible change.

Maximum Click Length

This parameter controls the maximum click lengt. It can be set by pressing the "click type" key, selecting "CLK-LENGTH" (key B) and adjusting the value with the "data" encoder. Clicks longer than this value will be ignored. This parameter should be set to the lowest value allowing removal of all clicks. A too high value will require excess DSP power and hence reduce interpolation quality somewhat, though the effect will be quite subtle.

For most typical work the 100 samples setting will suffice. However large scratches resulting from serious gramophone surface damage usually result in very large clicks and therefore require a higher maximum click length value.

Smooth

Small, single sample ticks in the audio can be removed by engaging the "Smooth" option. Smooth can be adjusted by pressing the "quality" key and toggling the "Smooth" option with key A (see Fig. 2). In addition to removing small clicks this will attenuate high frequency noise (see Fig. 12).
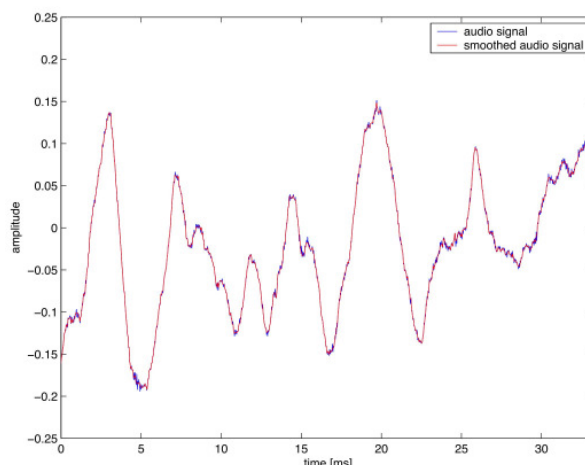
Fig. 12: Smooth effect

Click Monitor

If the click monitor is switched on you can listen to the difference of the input signal and the restored signal, i.e. the removed

clicks. The click monitor can be switched on and off by pressing the "quality" key and toggling the "monitor" option with key B (see Fig. 2). This will be helpful for adjusting the parameters as for certain source materials it is easier to hear the effect of the different parameter values.

Latency

The DeClicker latency can be adjusted over a range of 0.15 to 3.00 seconds. It can be set by pressing the "quality" key, selecting "LATENCY" (key D) and adjusting the value with the "data" encoder. Higher latencies are basically preferred as they give less variation in interpolation quality. On the other hand, short latencies greatly help while setting parameter values. That's because a parameter value change will become audible only after the latency time (e.g. you'll need to wait 3 seconds in the highest latency setting until you hear the effect of a parameter change).

Usually it will be convenient to use a medium or low latency (say around 0.50 seconds) for setting parameters and a high latency (above 1.00 seconds) for the final restoration pass.

Quality Factor ("Q")

The quality factor is displayed in ❽. It is a measure for the interpolation quality achieved by the DNA1. The interpolation quality is adaptive to the workload (number and size of the clicks) to make full use of the DSP power all the time and to provide best restoration quality. The quality factor value ranges from 0 to 100. Higher values indicate better interpolation quality.

The variation of the quality factor over time is dependent on the latency setting. Large latency will provide less variation of the quality factor over time. If the quality factor reaches zero new clicks are discarded in order to avoid further reduction of the interpolation quality. Use a multi-pass restoration to handle these cases.

Multi-Pass Restoration

As mentioned above, the quality of the interpolation process is adapted to the current workload. If there are many clicks, you may consider the interpolation quality to be insufficient in critical cases or the declicker might even discard some clicks if the quality factor reaches zero. A multi-pass restoration will distribute the number of clicks amongst two or more passes (removing the largest clicks first) and hence allow for a higher interpolation quality. The suggested approach is as follows:

1. Set the latency of the DNA1 to a medium or low value to simplify adjustment of the parameters.

2. Set all parameters such that all audible clicks are removed.

3. Set the latency to a high value for a constant quality factor.

4. Increase the detection threshold until the resulting quality factor is sufficiently high and make the first pass.

5. For the following pass(es), decrease the detection threshold as much as possible while maintaining the desired quality factor until you reach the original detection threshold.

For cases where there are very different click types (e.g. large scratches and fine crackles) a multi-pass restoration has the aditional advantage of allowing differing settings for each pass (e.g. higher asymmetry for the large scratches). Large clicks must always be removed first.

Strategy For Setting The Parameters– Method I (quick and dirty)

1. Choose the factory preset which best fits the clicks and material at hand.

2. Set the detection threshold to the highest value which detects all clicks.

3. Set the extension threshold to the highest value which allows clean removal of all clicks.

Strategy For Setting The Parameters–Method II (less quick and less dirty)

1. Set the detection threshold to infinity, the extension threshold to 2.0 and the discard threshold to 0.0

2. Set the adaptivity according to the click size and dynamic behaviour of the music (see section Adaptivity for details).

3. Select an appropriate asymmetry start value according to the click type and shape at hand (see section Asymmetry for details).

4. Decrease the detection threshold to the highest value which detects all clicks.

5. Set the extension threshold to the highest value which allows clean removal of all clicks.

6. Increase the discard threshold to the highest value which still detects all clicks.

7. Repeat steps 4 and 5 with different asymmetry and adaptivity settings to see if you can improve the result.

In many cases, step 7 may not be necessary.

Denoiser

The "on" key in the Denoiser section has three states: on, hard bypass and soft bypass. If the "on" key is not lit, the whole Denoiser block is skipped in this case and the Denoiser latency is also taken out of the signal chain (hard bypass mode). A short push on the "on" key will enable the Denoiser section and the key will light solidly (on mode).

Another short push will set the Denoiser section in soft bypass mode where the audio signal is not denoised but the latency remains. The soft bypass mode is indicated by a flashing "on" key.

To activate the hard bypass again press the "on" key for about 2 seconds.

There are three different modes of operation for the denoiser: auto, semi-auto and manual.

> As a general rule of thumb, the auto or semi-auto mode should be applied to signals with varying signal-to-noise ratio, while the manual mode is very robust for denoising constant SNR or extremely low SNR input.

Auto Mode

This is the easiest choice for removing noise and can provide very good results for most types of noisy signal. Once this mode is selected, the algorithm works autonomously and cannot be tuned by the user.

Three levels of noise reduction are possible in auto-mode:

Low: Use this setting when there is low level noise / hiss on the track, for example when transferring a high quality analogue tape master to digital format.

Mid: This is useful for more noisy analogue tape recordings, where there is significant hiss or other noise artefacts present.

High: The automatic High setting is needed when dealing with noisy material containing significant wideband noise.

Semi-Auto Mode

When switching from auto mode to semi-auto mode, the current workspace (A, B, CH1 or CH2, depending on the "A-B" and the "ch1" or "ch2" keys) is initialized, such that the behaviour of the denoiser is exactly the same as in the auto mode specified by the auto intensity setting ("high", "mid" or "low").

If one switches between auto mode and manual mode (not using the semi-auto mode), the workspace is retained.

> To compare a manual mode setting to a semi-auto mode setting, it is necessary that the manual mode workspace be stored as a snapshot! Otherwise the manual mode setting is lost when switching to semi-auto mode.

In semi-auto mode, the same parameters for fine-tuning are available as in manual mode, except for the "SNR"–this is replaced by the parameter "Intensity", which controls the amount of noise reduction (0% results in slight noise reduction, 100% in heavy noise reduction). Thus semi-auto uses the same adaptive SNR detection as the auto-mode, but with adjustable parameters.

Manual Mode

To explain the manual set-up of the Denoiser, a quick overview of its inner workings are in order.

The signal is segmented into overlapping chunks of audio (called frames) and analyzed by the Intelligent Analyzer sidechain which evaluates the information of the signal in time and frequency domain and controls the Noise Separator section.
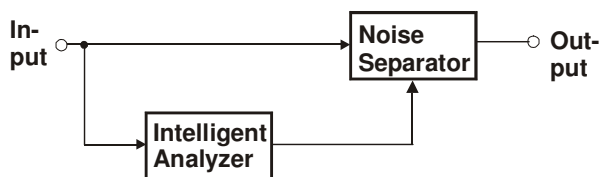
Fig. 13 Denoiser Block Schematic

The noise separator block is a linear-phase convolution engine. The convolution order is determined by the frequency resolution parameter "FFT RES.": the higher the frequency resolution, the higher the convolution order. A high-order convolution engine is necessary for removing noise in material with dense harmonic content.

The Intelligent Analyzer performs a frequency decomposition on each frame (the length of the frame is also determined by the "FFT RES." parameter). The magnitude of the frequency decomposition (spectrogram) is shown in the display (see Fig. 14).
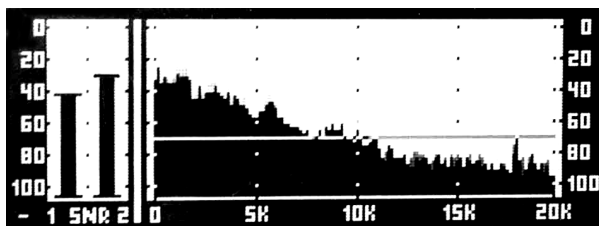
Fig. 14: Denoiser Display Section

Based on the setting of the "SNR" parameter (in manual mode), or on the auto mode level, it estimates a noise threshold (shown as a solid horizontal line in the display). This is a first distinction between signal and noise (components below the threshold are considered noise).

Using the threshold, timing specifications (see below) and other parameters, the Intelligent Analyzer configures the Noise Separator. The Noise Separator is a freely programmable convolution engine, and the Intelligent Analyzer supplies the convolution coefficients.

The Noise Separator will *not* remove everything below the threshold line–how much is effectively removed depends further on the "ambiance level" as well as the timing specifications (see below).

The most important parameters for manual set-up of the DNA1 are "SNR" and "ambience" (the two rotary knobs in ❹).

By adjusting "SNR", one specifies the signal-to-noise ratio (SNR). This parameter is crucial–if set too low, noise will intermittently break through the noise separator, resulting in unpleasant noise modulation. If set too high, parts of the programme will incorrectly be regarded as noise and removed (ambiance or even musical content). A helpful feedback while adjusting the "SNR" is the threshold line and the estimated SNR located in display section ❽ and labelled "SNR" (if labelled "BUF", touch any knob in the Denoiser section ❹ to switch to the SNR meters). The thin bars represent the SNR, while the thick bars on top of the thin bars represent the SNR improvement due to the Noise Separator.

Usually the SNR does not stay constant throughout the whole programme. For example during a soft part in a song, the SNR might decrease. In that case, to achieve optimal results, the DNA1 has to be set-up for each part separately (in auto mode, fluctuations of the SNR are compensated for).

Once the "SNR" has been set correctly, the amount of noise reduction can be adjusted with the "ambience level". It is usually simplest to set-up the "SNR" with the "ambience" set to "off", which means that all of the noise is removed (this way it is easier to determine the correct setting for the "SNR"). By increasing the "ambience", the noise separator permits increasing amounts of ambience (and noise) to seep back into the signal. A setting of 0.0 dB is equivalent to no noise reduction.

After an initial set-up of these two parameters, they can then be iteratively fine-tuned for optimal results.

Flip

The Flip button flips (or swaps) the desired signal and noise. This means you are asking the DNA1 to separate and output the "noise" rather than the "signal". This is useful in cer-

tain cases where the desired part is the much weaker component of the signal.

To get best results with the flip mode, use very fast timing constants for the bands above the noise discrimination threshold line. As bands below the threshold line are not affected in the flip mode, their timing constants don't matter.

> **Both the analysis and separator functions are swapped. This is not equivalent to the original signal minus the noise-processed signal. A new analysis is performed and therefore, this function cannot reliably be used to judge the quality of any settings by flipping and listening to the noise component.**

Most content can satisfactorily be improved using the above-described parameters. However, to also be able to process tracks with very wide dynamic range, extremely high noise levels, or non-standard noise types, many more parameters are available to fine-tune and adapt the DNA1 to these situations. Their effect and set-up will be described below.

Advanced Settings

Frequency Resolution

By selecting "menu" "Denoiser" one can adjust the frequency resolution of both the analyzer and the separator section. Five different settings are available, from "128" to "2048". These are the number of frequency points–a higher number means a higher resolution in the frequency domain. High frequency resolution comes at the cost of a longer overall delay:

Frequency Resolution	Denoiser Delay
2048	320 ms
1024	160 ms
512	85 ms
256	48 ms
128	27 ms

Table 1: Denoiser Delay vs Frequency Resolution

Generally, the lower the frequency resolution the faster the update rate of the Noise Separator. Thus fast changing noise profiles or highly transient material should be processed at a low frequency resolution. The drawback is that the Noise Separator also has a lower frequency resolution, thus the signal can be degraded with a higher probability.

An optimal setting (if it exists) is very programme dependent. Usually 512 or 1024 points are a good compromise. Heavily corrupted speech can sometimes be more effectively processed at 128 points.

Averaging

Averaging determines how many frames are averaged together and then analysed (and correspondingly, how fast the Noise Separator is updated). A setting of "1/1" means that every frame is analysed, one has maximum overlap of consecutive frames. A setting of "1/2" means that every second frame is analysed, and so on. The default setting is "1/1". For a setting of "1/k", where (k–1) is the number of omitted frames, k spectrograms are averaged together to get a more accurate noise profile.

This trades off a better noise estimate with slower adaptivity of the Noise Separator.

Crossover

The Intelligent Analyzer splits the frequency range into three bands: low, mid and high. Certain parameters, described below, can be adjusted separately for each band.

For example, one might be faced with the problem of denoising an old music recording.

With the crossover, one can divide the signal into three different bands: components above 12 kHz are likely to be hiss, between 5 kHz and 12 kHz is a transition region, and most of the signal energy resides below 5 kHz.

Another example might be forensic de-noising of speech signals. Then most signal information is contained in the range from 400 Hz to about 6 kHz, and the two bands outside this range can be considered as noise.

This is a priori information known about the input signal, and we want to exploit it for the denoiser (by setting the filter timing appropriately, see below).

To set-up the crossover, enter the Denoiser sub menu by touching one of the Denoiser rotary knobs ("SNR Improvement" or "ambience level"). Press key A to enter the crossover submenu. One can now adjust the two crossover frequencies splitting the signal into three bands. By pressing key C, one can adjust the mid / high crossover frequency, and by pressing key D, one can adjust the low / mid frequency with the "data" rotary knob.

Preshape

In some cases, where the noise has certain dominant parts in the frequency domain, the preshape function can be used to reduce the energy in that band before analysis takes place and thus increase the probability of the noise being detected by the Intelligent Analyzer.

In other cases, where a certain band has desirable but very weak components in a generally noisy situation, the preshape function can be deployed to enhance the likelihood of the weak signals to be untouched by the DNA1.

Note: this is not the same as using an equaliser to pre-colour the track. An EQ will change the frequency characteristics of the output since it filters the incoming signal. In the DNA1 the preshaping is only used in the analysis sub-section and a clean, unweighted version is then sent to the separator sub-section.

To adjust the preshape, enter the Denoiser submenu by touching one of the Denoiser rotary knobs ("SNR" or "ambiance"). Press key A to enter the crossover submenu. Press-ing key B allows adjusting the preshape parameter:

- "Vari." means that each of the three bands can be weighted separately (as described below),

- "A-Wght" applies the standard A-weighting curve,

- "off" disables preshaping.

Crossover Band Parameters

One can view the Noise Separator convolution engine as a multiband noise gate (with up to 1025 gates running in parallel). The noise gate uses parameters threshold (which is adaptively determined each frame for all gates simultaneously), suppression (determined by the "ambiance"), attack and release (see Fig. 15: Noise Gate Dynamics Transfer Curve).

Instead of manually having to set-up each noise gate one by one, they are grouped together into the three bands specified by the crossover frequencies.
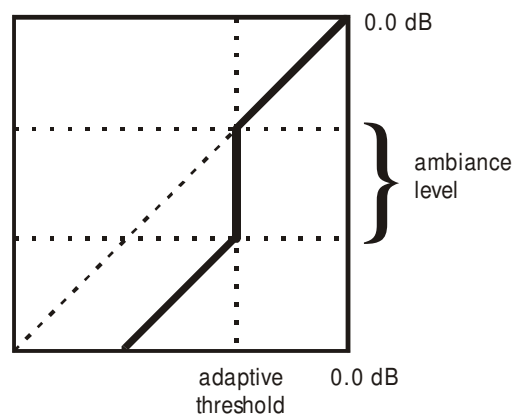

Fig. 15: Noise Gate Dynamics Transfer Curve

Attack

This is the attack time of the noise gate (measured in frames). If a band contains critical material, one wants the gates to act conservative. This is achieved by using a longer attack rate. For bands that are considered to contain more noise, the attack rate can be set faster for more efficient noise removal.

Release

The release time can be adjusted with similar considerations as for the attack time.

> **When the frequency resolution is adjusted, the frame time changes and thus also the absolute attack and release times!**

Weight

Similar to the A-weighting function, each band can be issued with its own weight. Weighting is a way to help the Intelligent Analyzer classify noise and signal. If a band is known to contain more noise than signal, but this noise is of high energy, the Analyzer will not recognize it as such. By weighting the band down, the noise energy in the analysis process is reduced, and thus the probability of noise reduction raised.

The applied weighting is visible in the real time analyzer display.

Weighting is an aid for the analysis process and does not affect the signal as such.

Tips on adjusting the Denoiser processor

1.) Between the "auto" and the "semi-auto" mode there is some interaction.

First select the auto mode and one of the three settings low/mid/high. After that select the semi-auto mode. This causes the settings of the selected auto mode to be copied into the parameters of the semi-auto mode, i.e. the auto mode sets the starting point for the semi-auto mode.

Consequently when you switch back to auto mode, the semi-auto mode settings are reset to the appropriate auto mode settings, so make sure to save the semi-auto mode settings before you switch back to auto mode, provided you would like to keep them.

The manual mode on the other hand is fully independent from the auto/semi-auto modes.

2.) For very noisy signals we recommend to use a low "FFT RES." value, while for signals with low noise level higher "FFT RES." values are better. "FFT RES." parameter: press the "menu"   "Denoiser".

3.) When using the DNA1 in unganged mode, we recommend to copy the CH1 parameters into the CH2 parameters before switching off the ganged mode. This makes sure that both channels have the same parameter values to start with. Press the "copy" key to get to the copy menu.

4.) For signals with a relatively low noise level the M/S mode is very useful. Switch to M/S with the "M/S" key, then copy CH1 to CH2 (see 3.) above) and then switch off the "ganged" key. Now apply the proper denoising to the M channel only. In the S channel do not apply any denoising or only a small amount. This leaves the ambience (reverb) information in the S channel intact and still takes out the noise that often is strongest in the M channel.

5.) There are cases where the noise in one channel is higher than in the other channel. In such a case we recommend to switch the DNA1 to unganged mode and treat the two channels separately. It may even be advisable to use different modes in the two channels (auto/semi-auto/manual). The "FFT RES." parameter is the same for both channels at all times.

Noise-Reduction Controls

See Fig. 1 Front Panel and Display Sections, ❷. These keys affect the Denoiser and the Declicker controls.

M/S

In the M/S domain, the two channels representing the stereo signal contain information on the Mid (mono) and Side content of the stereo signal. L/R and M/S format can be interchanged without signal degradation.

Denoising in the M/S domain can be interesting for highly correlated stereo signals to remove hiss and noise in the S channel without affecting the main part of the signal in the M channel or vice versa.

> **To take advantage of the M/S feature, the controls have to be unganged!**

Toggling the "M/S" key activates the MS mode.

CH1/M & CH2/S

These two keys patch the denoiser and Declicker front panel controls to the according channels. The controls can either be separate for each channel or ganged together. The key is lit for the active channel, e.g. "ch1" key is lit means that all front panel contols affect channel 1 processing.

To copy CH1 settings to CH2 or vice versa, press the "copy" key and select an option (Fig. 3: "copy" menu).

The displayed real-time analysis depends on the channel setting. If CH1/M is selected, the channel 1 (or mono) signal is displayed, if CH2/S is selected, then the channel 2 (or side) signal is displayed.

Ganged

When this key is lit, the band controls act for both channels simultaneously (so it does not matter which key is lit, "ch1" or "ch2"). Switching channels while the "ganged" key is lit will activate CH1 and CH2 settings alternately for both channels (both settings are stored separately).

If ganged is off, the controls will affect the active channel only.

K-Stereo (by Bob Katz)

In the factory preset snapshot, the K-Stereo processing is already set for a good starting point.

The controls for the K-Stereo processor are found in section ❻ of the DNA1 front panel.

Adjust the K-Level (left) knob, in 0.5 dB increments, to affect the degree of ambience extraction and stereoization. 0 dB is a good level to start with. Delicate or intimate musical material requires less K-level, poorly recorded material, or special effects require more.

What does a K-level of "0 dB" mean? This is strictly a calibrated start point. It is the most common K-Level to start at. There is absolutely no problem using a "minus" or a "plus" setting as well. A setting of K = −1 dB simply means 1 dB less extracted ambience than the reference of 0. Of course, it becomes more and more subtle the more you turn it down. Most people give up after −3 dB and it's real hard to hear an enhancement below about −6 dB with most material.

The limit of K-Level is +9 dB. It is extremely rare to desire a K-Level setting greater than +2 dB, especially with the "deep" option active.

As a rule of thumb, you will need 1 or 2 dB less K-level when activating the "deep" option. Nevertheless, don't be afraid to use higher K-levels if the material calls for it; just watch out for that "too big" sound. Instruments will take on a gorgeous sense of depth, stereophonic life and definition. Here's a secret: K-Stereo makes the sound louder, but rarely causes clipping (shhhh).

If both "wide" and "deep" options are active, the algorithm is most powerful, with the best ambience extraction, and the widest and deepest soundstage.

If you want less of an effect, try lowering the K-Level, and/or changing the "wide" and "deep" options. If a recording is already quite spacious width-wise, or if you do not want to make it more spacious width-wise, then try to set the "wide" setting off to increase the depth from front to back.

Ambiance Filters

With the DNA1 K-Stereo, the mastering engineer can equalize the ambience in a recording. The most commonly used ambience EQ you will use is the High-Pass filter, which is a first order "bass cut" or "high pass" filter *on the extracted ambience only*.

With this control, it's possible to tighten the spatiality of the bass instrument(s) without affecting the ambience in the midrange and highs.

We've provided two other ambience filters, a low-pass and a bell. These will be rarely used. We've occasionally used the low-pass filter when material comes in that sounds harsh and bright and we want to prevent this distortion from "leaking" into the ambience.

Since the ambience extraction algorithm depends on wide bandwidth, there is a point of diminishing returns.

We've also provided a bell (parametric) filter on the ambience, which you may use occasionally, for example, to warm up the ambience. This can be powerful when working with poorly recorded stereo masters. For example, we've successfully used the parametric EQ in a situation where a vocalist sounded just fine and didn't need any warming up, but the instruments were a bit bright and harsh.

Standard equalization techniques don't work in this situation. But with K-Stereo, you can selectively equalize the ambience or reverberation in a recording. You may not be able to turn a sow's ear into a silk purse every time, but this tool can turn quite a few tricks.

Patching Recommendations

We recommend patching this processor early in the digital signal chain, in front of any compressors or other dynamics processors. Patch the K-Stereo between a 2-channel reproducer and a digital console,

or in a console's digital insert point. You can also use an external A/D/A converter around the K-Stereo box for use with an analog system. Use the highest quality A/D/A for this purpose (ideally, 96 kHz/24 bit).

Tips on adjusting the K-Level Processor

Adjust the K-level to your aesthetics. Take advantage of the "on" key to test if you have "spaced out" too much. In most cases, you will discover you want a little less K-level than you had during that initial flush of excitement! Remember to keep a proper balance amongst clarity, intimacy, and spaciousness (which equals "power and depth").

K-levels below about –5 dB are extremely subtle and will rarely be used. –10 is effectively OFF to all but the most sensitive ear.

Output Control

The Output Control section ❼ of the DNA1 front panel contains five rotary knobs and two switches.
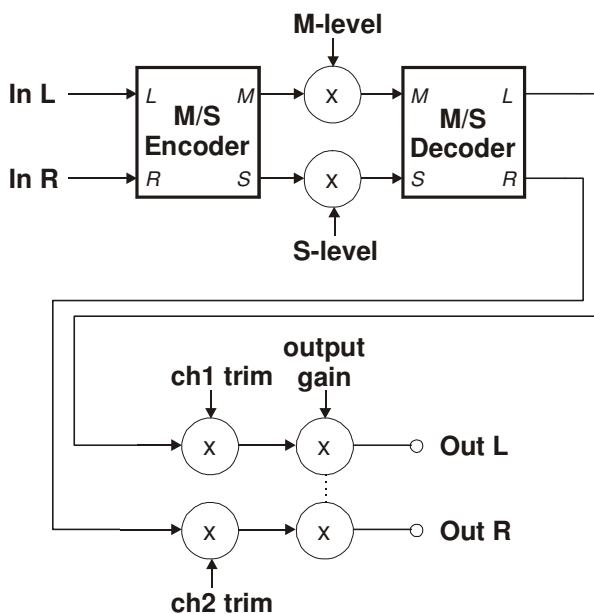
Fig. 16: Output Control Section

"M-level" and "S-level" control the gain of the mid-channels and side-channels after an M/S matrix encoding of the signal. If these levels are off (i.e. either the "on" key is not lit or both gains are set to 0.0 dB) the

M/S encode and decoder are bit transparent.

"ch1-trim" and "ch2-trim" allow separate control of both channel gains while the "output gain" applies a combined gain to both channels.

The "on" key in the trim section affects only the trims, not the output gain.

All gain controls are bit transparent bypassed with a setting of 0.0 dB.

Even with extreme volume differences snapshot comparing causes no audible signal discontinuities (clicks) due to the gain sections. Special care has been taken to guarantee soft gain transients.

Dither

The DNA1 is able to apply POW-r type dither algorithms to avoid distortion when re-quantizing from 24 bit to 16 bit or 20 bit output word length (see TECHNICAL DATA for dither specifications).

To toggle dither on/off, press key D twice. To see dither status refer to the status display (see Fig. 2: Status Display).

Word Length

To set output word length, press key D and select the appropriate word length.

Setup

Press key D then chose the setup menu. This leads to the menu where the type of POW-r dither can be chosen (see TECHNICAL DATA for dither specifications). Also selectable is:

Auto-Blacking

If auto-blacking is activated by marking the "0 IN   0 OUT" option, dither will be turned off if the input signal is zero As soon as the input signal is non-zero, dither will be turned on again. This ensures that pauses between programmes are still digital zero, even if dither is activated.

The name "black" for zero comes from digital image processing, where a black pixel has value zero.

Spectrum Analyzer Magnitude Resolution

To obtain the maximum benefit from the real-time spectrum analyzer, set the vertical display resolution to a range that covers the input signal. There are four ranges available:

0 … –40 dB,

0 … –60 dB,

0 … –80 dB,

0 … –100 dB,

The optimal range will depend on the absolute level and the SNR of the input signal. To change the display resolution, press the "menu" "system" "Display". Choose a range and press the "menu" key again to return to the status display.

LCD Contrast

To change the display contrast, press the "menu" key, and then select "Display". Adjust contrast with the "gain/data" encoder.

Bypass

The DNA1 can be completely bypassed with the green "bypass" key below the mains switch. If the "bypass" key is lit, the DNA1 is absolutely transparent, making the output signal bit equal to the input signal. The signal passes through the DNA1 with a minimal latency.

Remote Control

All workspace parameters as well as control settings can be controlled externally.

To see remote control status refer to the status display.

Protocol

To select a remote control protocol, press key C from the status display menu. Select either "RS 232", "RS 422" or "MIDI". This will determine which port at the rear of the DNA1 is active. The other inactive ports will be ignored.

Press the "menu" key to return to the status display if you do not want to change remote on/ off status.

When selecting a port other than MIDI you have to specify the baud rate. Refer to the remote control software manual for correct settings. The set baud rate is displayed to ensure proper set-up.

Channel

To change the remote control channel number, press key C (adjacent to the remote control field). Use the "gain/ data" encoder to select a value between 1 and 16. Press the "menu" key to return to the status display.

The DNA1 does not support poly-mode (listening on all channels). Select the same channel on the controller software as is set internally, else no remote control messages will be received.

On/Off

To enable or disable remote control, press key C (adjacent to the remote control field). Press it again to toggle the "ON/OFF"-value.

While remote control is off, data will still be sent to the selected port. Also, the MIDI THRU output always mirrors the MIDI IN input data regardless of "ON/OFF" or protocol status.

Self Test

Press the "menu" key, then select the "System"   "Selftest" option. This will start a procedure that checks the functionality of all input and display devices.

Auto Touch Menu Switchback

All display menus that are activated by the touch of a knob from sections ❹–❼ automatically revert to the status display after the touch has been released for a couple of seconds. This auto switch back feature can be turned off by pressing the "menu" key, then selecting the "System"   "Controls" menu. Check or uncheck the checkbox by pressing key B.

Software Information

Press the "menu" key, then select the "System"   "About" option. This displays a screen with all the relevant information on your DNA1 and how to contact us. Have this page ready when you want to report a problem.

TECHNICAL DATA

Hardware

DSP performance:	900 MFLOPS, 32/40 bit floating-point
DSP memory:	340 kB Cache, 16 MB SDRAM
Host processor:	25 MHz, 16 bit
FLASH memory:	256 kB

AES/EBU Input

Sampling Frequencies:	44.1 kHz, 48.0 kHz, 88.2 kHz, 96.0 kHz
Maximum Input Word length:	24 bits
Channel Status Data:	Input accepts professional or consumer format.
Channel Status Bits forwarded to AES/EBU output:	see table below
Connector:	XLR female

AES/EBU Output

Sampling Frequencies:	44.1 kHz, 48.0 kHz, 88.2 kHz, 96.0 kHz (always the same as the input)
Output Word length:	24 bits
Connector:	XLR male

AES/EBU Channel Status Data

The DNA1 allows converting the incoming Channel Status Data as follows:

- From Consumer to Professional

- From Professional to Consumer

- Transparent mode, i.e. Channel Status Data is fed forward to the output without any conversion (one exception, see below)

The following tables describe how the Channel Status Data bits are generated in the various conversion modes.

Output selected: Consumer format.

> *Input: Consumer format.*
> Output: All bits fed forward (transparent), except for:
> Byte 1:   Bits 0…6: 0000000 (category code general)
>   Bit 7: 1 (original)
>
> *Input: Professional format.*
> Output:
> Byte 0:   Bit 0: 0 (consumer)
>   Bit 1: 0 (audio)

Bit 2: 1 (copy allowed)

Bits 3,4: Preemphasis according to input

Bit 5: 0 (2 channel mode)

Bits 6, 7: 00 (Mode 0)

Byte 1:   Bits 0…6: 0000000 (category code general)

Bit 7: 1 (original)

Byte 2:   Bits 0,1,2,3: Sampling Frequency according to input

Bits 4, 5: 00 (accuaracy grade II)

Bits 6, 7: 00

Bytes 3…23:   reserved bytes

Output selected: Professional format.

Input: Professional format.

Output: All bits fed forward (transparent), except for:

Byte 2:   Bits 0, 1, 2: 001 (max. sample length = 24bit)

Bits 3, 4, 5: 101 (24 bit word length)

Bits 6, 7: 00

Input: Consumer format.

Output:

Byte 0:   Bit 0: 1 (professional)

Bit 1: 0 (audio)

Bits 2, 3, 4: Preemphasis according to input

Bit 5: 0 (source fs locked)

Bits 6, 7: Sampling Frequency according to input

Byte 1:   Bits 0, 1, 2, 3: 0001 (two channel mode)

Bits 4, 5, 6, 7: 0000 (no user bit encoding)

Byte 2:   Bits 0, 1, 2: 001 (max. sample length = 24bit)

Bits 3, 4, 5: 101 (24 bit word length)

Bits 6, 7: 00

Bytes 3…12: All bits 0

Byte 23:   CRCC byte

Output selected: Transparent

Input: Any format.

Output: All bits fed forward (transparent), except if not bypassed:

Byte 2:   Bits 0, 1, 2: 001 (max. sample length = 24bit)

Bits 3, 4, 5: 101 (24 bit word length)

Bits 6, 7: 00

Power

Mains Voltage: 110/220 Volts with voltage selector
Fuse rating: 500 mA slow blow
Power Consumption: 40 VA max

Overload

Number of consecutive
over-samples to cause
"over" display: 1…16 settable

Parameter Table

Denoiser SNR improvement

0.0, 0.5, 1.0, 1.5, 2.0, 2.5, 3.0, 3.5, 4.0, 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, 9.5, 10.0, 10.5, 11.0,
11.5, 12.0, 12.5, 13.0, 13.5, 14.0, 14.5, 15.0, 15.5, 16.0, 16.5, 17.0, 17.5, 18.0, 18.5, 19.0, 19.5, 20.0,
20.5, 21.0, 21.5, 22.0, 22.5, 23.0, 23.5, 24.0, 24.5, 25.0, 25.5, 26.0, 26.5, 27.0, 27.5, 28.0, 28.5, 29.0,
29.5, 30.0, 30.5, 31.0, 31.5, 32.0, 32.5, 33.0, 33.5, 34.0, 34.5, 35.0, 35.5, 36.0, 36.5, 37.0, 37.5, 38.0,
38.5, 39.0, 39.5, 40.0, 40.5, 41.0, 41.5, 42.0, 42.5, 43.0, 43.5, 44.0, 44.5, 45.0, 45.5, 46.0, 46.5, 47.0,
47.5, 48.0, 48.5, 49.0, 49.5, 50.0, 50.5, 51.0, 51.5, 52.0, 52.5, 53.0, 53.5, 54.0, 54.5, 55.0, 55.5, 56.0,
56.5, 57.0, 57.5, 58.0, 58.5, 59.0, 59.5, 60.0, 60.5, 61.0, 61.5, 62.0, 62.5, 63.0

Denoiser Intensity [%]

0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 30,
31, 32, 33, 34, 35, 36, 37, 38, 39, 40, 41, 42, 43, 44, 45, 46, 47, 48, 49, 50, 51, 52, 53, 54, 55, 56, 57, 58,
59, 60, 61, 62, 63, 64, 65, 66, 67, 68, 69, 70, 71, 72, 73, 74, 75, 76, 77, 78, 79, 80, 81, 82, 83, 84, 85, 86,
87, 88, 89, 90, 91, 92, 93, 94, 95, 96, 97, 98, 99, 100

Denoiser ambiance level

0.0, 0.5, 1.0, 1.5, 2.0, 2.5, 3.0, 3.5, 4.0, 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, 9.5, 10.0, 10.5, 11.0,
11.5, 12.0, 12.5, 13.0, 13.5, 14.0, 14.5, 15.0, 15.5, 16.0, 16.5, 17.0, 17.5, 18.0, 18.5, 19.0, 19.5, 20.0,
20.5, 21.0, 21.5, 22.0, 22.5, 23.0, 23.5, 24.0, 24.5, 25.0, 25.5, 26.0, 26.5, 27.0, 27.5, 28.0, 28.5, 29.0,
29.5, 30.0, 30.5, 31.0, 31.5, 32.0, 32.5, 33.0, 33.5, 34.0, 34.5, 35.0, 35.5, 36.0, 36.5, 37.0, 37.5, 38.0,
38.5, 39.0, 39.5, 40.0, 40.5, 41.0, 41.5, 42.0, 42.5, 43.0, 43.5, 44.0, 44.5, 45.0, 45.5, 46.0, 46.5, 47.0,
47.5, 48.0, 48.5, 49.0, 49.5, 50.0, 50.5, 51.0, 51.5, 52.0, 52.5, 53.0, 53.5, 54.0, 54.5, 55.0, 55.5, 56.0,
56.5, 57.0, 57.5, 58.0, 58.5, 59.0, 59.5, 60.0, 60.5, 61.0, 61.5, 62.0, 62.5, 63.0

Denoiser frequency resolution [number of frequency points]

128, 256, 512, 1024, 2048

Denoiser crossover frequencies [Hz]

100, 257, 413, 570, 727, 883, 1040, 1197, 1354, 1510, 1667, 1824, 1980, 2137, 2294, 2450, 2607, 2764,
2920, 3077, 3234, 3391, 3547, 3704, 3861, 4017, 4174, 4331, 4487, 4644, 4801, 4957, 5114, 5271, 5428,

5584, 5741, 5898, 6054, 6211, 6368, 6524, 6681, 6838, 6994, 7151, 7308, 7465, 7621, 7778, 7935, 8091, 8248, 8405, 8561, 8718, 8875, 9031, 9188, 9345, 9502, 9658, 9815, 9972, 10128, 10285, 10442, 10598, 10755, 10912, 11069, 11225, 11382, 11539, 11695, 11852, 12009, 12165, 12322, 12479, 12635, 12792, 12949, 13106, 13262, 13419, 13576, 13732, 13889, 14046, 14202, 14359, 14516, 14672, 14829, 14986, 15143, 15299, 15456, 15613, 15769, 15926, 16083, 16239, 16396, 16553, 16709, 16866, 17023, 17180, 17336, 17493, 17650, 17806, 17963, 18120, 18276, 18433, 18590, 18746, 18903, 19060, 19217, 19373, 19530, 19687, 19843, 20000

Denoiser attack time [frames]

1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 30, 31, 32, 33, 34, 35, 36, 37, 38, 39, 40, 41, 42, 43, 44, 45, 46, 47, 48, 49, 50, 51, 52, 53, 54, 55, 56, 57

Denoiser release time constants and AVERAGE [frames]

1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 30, 31, 32, 33, 34, 35, 36, 37, 38, 39, 40, 41, 42, 43, 44, 45, 46, 47, 48, 49, 50, 51, 52, 53, 54, 55, 56, 57, 58, 59, 60, 61, 62, 63, 64, 65, 66, 67, 68, 69, 70, 71, 72, 73, 74, 75, 76, 77, 78, 79, 80, 81, 82, 83, 84, 85, 86, 87, 88, 89, 90, 91, 92, 93, 94, 95, 96, 97, 98, 99, 100, 101, 102, 103, 104, 105, 106, 107, 108, 109, 110, 111, 112, 113, 114, 115, 116, 117, 118, 119, 120, 121, 122, 123, 124, 125, 126, 127

Declicker SMOOTH

on/off

Declicker LATENCY [s]

0.15, 0.20, 0.25, 0.30, 0.35, 0.40, 0.45, 0.50, 0.60, 0.70, 0.80, 0.90, 1.00, 1.20, 1.40, 1.60, 1.80, 2.00, 2.20, 2.40, 2.60, 2.80, 3.00

Declicker DETECT. THR.

3.0, 3.1, 3.2, 3.3, 3.4, 3.5, 3.6, 3.7, 3.8, 3.9, 4.0, 4.2, 4.4, 4.6, 4.8, 5.0, 5.2, 5.4, 5.6, 5.8, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, 9.5, 10.0, 11.0, 12.0, 13.0, 14.0, 15.0, 16.0, 17.0, 18.0, 19.0, 20.0, 21.0, 22.0, 23.0, 24.0, 25.0, 26.0, 27.0, 28.0, 29.0, 30.0, 32.0, 34.0, 36.0, 38.0, 40.0, 45.0, 50.0, 55.0, 60.0, 70.0, 80.0, 90.0, 100.0, inf

Declicker EXTENS. THR.

0.5, 0.55, 0.6, 0.65, 0.7, 0.75, 0.8, 0.85, 0.9, 0.95, 1.0, 1.1, 1.2, 1.3, 1.4, 1.5, 1.6, 1.7, 1.8, 1.9, 2.0, 2.2, 2.4, 2.6, 2.8, 3.0, 3.2, 3.4, 3.6, 3.8, 4.0, 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, 9.5, 10.0

Declicker DISCARD THR.

0.00, 0.05, 0.10, 0.15, 0.20, 0.25, 0.30, 0.35, 0.40, 0.45, 0.50, 0.55, 0.60, 0.65, 0.70, 0.75, 0.80, 0.85, 0.90, 0.95, 1.00

Declicker CLK-LENGTH [samples]

50, 100, 200, 300

Declicker ADAPTIVITY

0.0, 0.1, 0.2, 0.3, 0.4, 0.5, 0.6, 0.7, 0.8, 0.9, 1.0

Declicker ASYMMETRY

0.0, 0.1, 0.2, 0.3, 0.4, 0.5, 0.6, 0.7, 0.8, 0.9, 1.0

Declicker Presets

Parameter:	Preset 1:	Preset 2:	Preset 3:	Preset 4:
Description	Small Clicks	Clicks	Large Clicks	Scratches
Detection threshold	16.0	20.0	24.0	40.0
Extension threshold	3.2	3.0	2.4	1.5
Asymmetry	0.2	0.4	0.6	0.9
Adaptivity	0.7	0.5	0.4	0.1
Discard threshold	0.05	0.05	0.1	0.3
Maximum click length	50	100	100	300

K-level [dB]

-17.0, -16.5, -16.0, -15.5, -15.0, -14.5, -14.0, -13.5, -13.0, -12.5, -12.0, -11.5, -11.0, -10.5, -10.0, -9.5, -9.0, -8.5, -8.0, -7.5, -7.0, -6.5, -6.0, -5.5, -5.0, -4.5, -4.0, -3.5, -3.0, -2.5, -2.0, -1.5, -1.0, -0.5, 0.0, 0.5, 1.0, 1.5, 2.0, 2.5, 3.0, 3.5, 4.0, 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0

K-Stereo Ambiance Filters Frequencies [Hz]

20 , 25, 31.5, 40, 50, 63, 80, 100, 126, 159, 200, 250, 315, 400, 500, 630, 800, 1000, 1260, 1590, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12600, 15900, 20000

K-Stereo Ambiance Filters Boost [dB]

-12.0, -11.5, -11.0, -10.5, -10.0, -9.5, -9.0, -8.5, -8.0, -7.5, -7.0, -6.5, -6.0, -5.5, -5.0, -4.5, -4.0, -3.5, -3.0, -2.5, -2.0, -1.5, -1.0, -0.5, 0.0, 0.5, 1.0, 1.5, 2.0, 2.5, 3.0, 3.5, 4.0, 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, 9.5, 10.0, 10.5, 11.0, 11.5, 12.0

K-Stereo Ambiance Q

0.4, 0.5, 0.6, 0.7, 0.8, 0.9, 1.0, 1.1, 1.2, 1.3, 1.4, 1.5, 1.6, 1.7, 1.8, 1.9, 2.0

Gains (M-level, S-level, ch1 trim, ch2 trim, output gain) [dB]

-∞, -92.0, -89.0, -86.0, -83.0, -80.0, -77.0, -74.0, -71.0, -68.0, -65.0, -62.0, -59.0, -56.0, -53.0, -50.0, -48.0, -46.0, -44.0, -42.0, -40.0, -39.0, -38.0, -37.0, -36.0, -35.0, -34.0, -33.0, -32.0, -30.0, -29.5, -29.0, -28.5, -28.0, -27.5, -27.0, -26.5, -26.0, -25.5, -25.0, -24.5, -24.0, -23.5, -23.0, -22.5, -22.0, -21.5, -21.0, -20.5, -20.0, -19.5, -19.0, -18.5, -18.0, -17.5, -17.0, -16.5, -16.0, -15.5, -15.0, -14.5, -14.0, -13.5, -13.0, -12.5, -12.0, -11.5, -11.0, -10.5, -10.0, -9.5, -9.0, -8.5, -8.0, -7.5, -7.0, -6.5, -6.0, -5.8, -5.6, -5.4, -5.2, -5.0, -4.8, -4.6, -4.4, -4.2, -4.0, -3.8, -3.6, -3.4, -3.2, -3.0, -2.8, -2.6, -2.4, -2.2, -2.0, -1.8, -1.6, -1.4, -1.2, -1.0, -0.8, -0.6, -0.4, -0.2, 0.0, 0.5, 1.0, 1.5, 2.0, 2.5, 3.0, 3.5, 4.0, 4.5, 5.0, 5.5, 6.0, 6.5, 7.0, 7.5, 8.0, 8.5, 9.0, 9.5, 10.0

Dither

Dithering is implemented using the POW-r set of algorithms. Below are the spectral characteristics of the three POW-r presets at 44.1kHz and 88.2kHz sampling rates each.
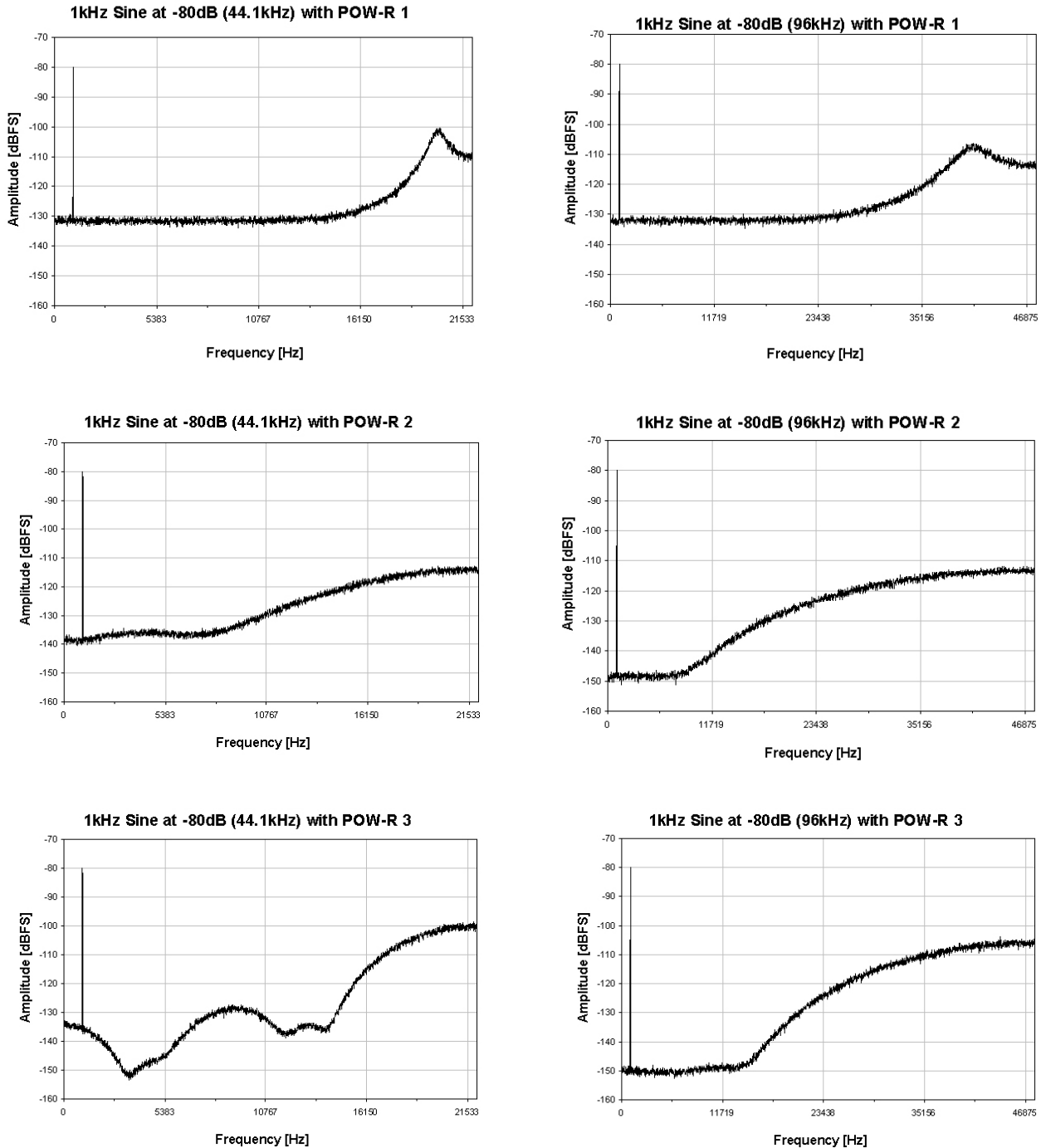

Fig. 17: Output spectrum with 80 dBFS/1 kHz sine wave, dithered to 16 Bits, using the three POW-r algorithms at 44.1 kHz and 88.2 kHz respectively

Remote Control Protocol

MIDI Implementation

Parameter Remote Control

All band parameters plus the controls parameters are remote controllable. Each parameter has its corresponding MIDI controller number.

According to the different parameters, the controllers are limited in their range. Invalid values are replaced by the maximum allowable values.

Following is a list of the supported MIDI system inclusive commands:

Action	MIDI Message	Status Byte	Data Byte(s)	Comment
Restore Snapshot	Program Change	1100 cccc	0ppp pppp	(cccc) Channel No. (ppppppp) New Prog No: 0 – 127
Denoiser CH1 SNR	Control Change	1011 cccc	00000000 ($00) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Denoiser CH1 Ambiance Level	Control Change	1011 cccc	00000001 ($01) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Denoiser CH1 Low Attack	Control Change	1011 cccc	00000010 ($02) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 56
Denoiser CH1 Mid Attack	Control Change	1011 cccc	00000011 ($03) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 56
Denoiser CH1 High Attack	Control Change	1011 cccc	00000100 ($04) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 56
Denoiser CH1 Low Release	Control Change	1011 cccc	00000101 ($05) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 126
Denoiser CH1 Mid Release	Control Change	1011 cccc	00000110 ($06) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 126
Denoiser CH1 High Release	Control Change	1011 cccc	00000111 ($07) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 126
Denoiser CH1 Low Vari. Weight	Control Change	1011 cccc	00001000 ($08) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 1 - 121
Denoiser CH1 Mid Vari. Weight	Control Change	1011 cccc	00001001 ($09) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 1 - 121
Denoiser CH1 High Vari. Weight	Control Change	1011 cccc	00001010 ($0A) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 1 - 121

Denoiser CH1 Xov　　Low/Mid	Control Change	1011 cccc	00001011 ($0B) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Denoiser CH1 Xov　　Mid/High	Control Change	1011 cccc	00001100 ($0C) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Denoiser CH1 Flip	Control Change	1011 cccc	00001101 ($0D) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
Denoiser CH1 On	Control Change	1011 cccc	00001110 ($0E) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1, 2
Declicker CH1 On	Control Change	1011 cccc	00001111 ($0F) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1, 2
Denoiser CH1 Preshape	Control Change	1011 cccc	00010000 ($10) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1, 2
Denoiser CH1 Intensity	Control Change	1011 cccc	00010001 ($11) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 100
Denoise CH1 low / mid/ high	Control Change	1011 cccc	00010010 ($12) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1, 2
Denoise CH1 auto/semi-auto/ manual	Control Change	1011 cccc	00010011 ($13) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1, 2
Declicker CH1 Presets	Control Change	1011 cccc	00010100 ($14) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 3
Declicker CH1 CLK-LENGTH	Control Change	1011 cccc	00010111 ($16) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 – 3
Declicker CH1 ADAPTIVITY	Control Change	1011 cccc	00011000 ($17) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 10
Declicker CH1 ASYMMETRY	Control Change	1011 cccc	00011001 ($18) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 10
Declicker CH1 EXTENS.THR.	Control Change	1011 cccc	00011010 ($19) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 42
Declicker CH1 DETECT.THR.	Control Change	1011 cccc	00011010 ($1A) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 60
Declicker CH1 DISCARD THR.	Control Change	1011 cccc	00011011 ($1B) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 20

Denoiser CH2 SNR	Control Change	1011 cccc	00100100 ($24) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
...	...	...	...	...
Declicker CH2 DISCARD THR.	Control Change	1011 cccc	01000011 ($3F) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 20
Declicker SMOOTH	Control Change	1011 cccc	01001000 ($48) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
Declicker LATENCY	Control Change	1011 cccc	01001001 ($49) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 57
Denoiser FFT RES.	Control Change	1011 cccc	01001010 ($4A) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 1,2,3,4,5
K-Stereo highpass	Control Change	1011 cccc	01001011 ($4B) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 31
K-Stereo lowpass	Control Change	1011 cccc	01001100 ($4C) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 1 - 32
K-Stereo peaking frequen-	Control Change	1011 cccc	01001101 ($4D) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 1 - 31
K-Stereo peaking boost	Control Change	1011 cccc	01001110 ($4E) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 48
K-Stereo peaking Q	Control Change	1011 cccc	01001111 ($4F) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 16
Output Control M-level	Control Change	1011 cccc	01010000 ($50) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Output Control S-level	Control Change	1011 cccc	01010001 ($51) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Output Control ch1 trim	Control Change	1011 cccc	01010010 ($52) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Output Control ch2 trim	Control Change	1011 cccc	01010011 ($53) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
Output Control output gain	Control Change	1011 cccc	01010100 ($54) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127
K-Stereo K-level	Control Change	1011 cccc	01010101 ($55) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 52
Denoiser AVERAGE	Control Change	1011 cccc	01010110 ($56) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 127

K-Stereo on	Control Change	1011 cccc	01010111 ($57) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
K-Stereo wide	Control Change	1011 cccc	01011000 ($58) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
K-Stereo deep	Control Change	1011 cccc	01011001 ($59) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
Output Control M/S on	Control Change	1011 cccc	01011101 ($5D) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
Output Control trim on	Control Change	1011 cccc	01011110 ($5E) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
nr control CH1 / 2 SELECT	Control Change	1011 cccc	01011111 ($5F) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
nr control ganged	Control Change	1011 cccc	01100000 ($60) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
nr control M/S	Control Change	1011 cccc	01100001 ($61) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
Declicker DECRACKLE DET.THRESH	Control Change	1011 cccc	01100010 ($62) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0 - 45
snapshots A - B	Control Change	1011 cccc	01100011 ($63) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
bypass	Control Change	1011 cccc	01100011 ($63) 0vvv vvvv	(cccc) Channel No. (vvvvvvv) New Value: 0, 1
System Reset		1111 1111	-	Recall Snapshot 0 to WS A

Dump Remote Control

All System-Exklusive commands consist of a header, a message type, possible data and the EOS (end of system exclusive). Below is the format of the header:

Byte	Value	Description	Section
0	F0	Sysex	Header
1	00	Manufacturer's ID	
2	30	Manufacturer's ID	
3	5A	Manufacturer's ID	
4	13	Model ID: 19 = Gambit DNA1	
5	0000nnnn	Device ID: -> Midi channel number	

Following is a list of the supported MIDI system exclusive commands (without header):

Snapshot format for dump. A complete dump consists of 128 snapshots:

Byte	Value	Description	Section
0-5	...	...	Header
6	00000001	message = Snapshot dump	Message Type
7	0nnnnnnn	Package number (0..7F)	Data
8	0nnnnnnn	scale set number	
9 – 107	0nnnnnnn	99 Bytes of Snapshot data	
108	0ccccccc	Checksum without byte one (F0 Sysex Command)	Checksum
109	F7	EOS	EOS

Dump request:

Byte	Value	Description	Section
0-5	...	...	Header
6	00000010	message = dump request	Message Type
7	F7	EOS	EOS

Version request:

Byte	Value	Description	Section
0-5	...	...	Header
6	00000011	message = version request	Message Type
7	F7	EOS	EOS

Version send:

Byte	Value	Description	Section
0-5	...	...	Header
6	00000100	message = send version	Message Type 4
7	0vvvvvvv	OS Version	data
8	0vvvvvvv	DSP Version	
9	0vvvvvvv	Scale Version	
10	0vvvvvvv	Coeff. Version	
11	F7	EOS	EOS

Workspace and Settings request:

Byte	Value	Description	Section
0-5	...	...	Header
6	00000110	message = send all parameters as single controler mes-	Message Type
7	F7	EOS	EOS

Programming Mode:

Note: This command triggers the programming mode in the DNA1. In this mode operation from the front panel is impossible!

Byte	Value	Description	Section
0-5	...	...	Header
6	00000101	message = enter programming mode	Message Type
7	F7	EOS	EOS

RS-232

The implementation is equal to MIDI.

Connector Type: D-SUB female, 9 pin, Pinout:

pin 1: n.c.

pin 2: RS232 output

pin 3: RS232 input

pin 4: connected to pins 6, 8

pin 5: ground

pin 6: connected to pins 4, 8

pin 7: n.c.

pin 8: connected to pins 4, 6

pin 9: n.c.

RS-422

The implementation is equal to MIDI.

Connector Type: D-SUB female, 9 pin, Pinout:

pin 1: remote input +

pin 2: remote output +

pin 3: n.c.

pin 4: ground

pin 5: ground

pin 6: remote input -

pin 7: remote output -

pin 8: ground

pin 9: ground