dCS 974

Digital to Digital Converter

User Manual

Software version 1.0x

May 2001

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1 ***dCS*** Ltd is Data Conversion Systems Ltd. Company registered in the England no. 2072115

PRODUCT OVERVIEW

The ***dCS 974*** DDC (Digital to Digital Converter) is a high performance real time sample rate and format converter, developed from our highly successful ***dCS 972***. It is designed for studio applications where source material is available in one format, but outputs are required in other digital formats in real time. For example, archives might be made for storage in 24/192 or 24/176.4 formats, and then used to produce output in SACD, DVD, CD and other multimedia formats. AES3, SPDIF, SDIF-2 and DSD formats are all supported, and multiple units may be synchronised for stable multi-channel operation.

The unit is mains powered and is housed in a 2U (3.5") high 19" rack mounting case. It may be controlled either from its front panel, or from a software based remote control running on a PC. Frequently used Setups may be stored and re-called later. The last setting is automatically stored on power down, so that fixed installations may be set up at leisure, installed and then left alone. Unauthorised alterations to settings may be prevented by a "panel lock out" feature.

Numerous monitoring functions are provided – both for the audio signal and for messaging attached to it. The unit has bit activity and level meters, and message manipulation. CRC, parity and invalid errors may be monitored and reported, so that "right first time" transfer to disc plants may easily be achieved.

The unit is highly software based, and more functions and features are added from time to time. Software updates from ***dCS*** are free![2]

Formats

- DSD at 2.822MS/s (see page **71**)
- PCM from 192 kS/s down to 11.025 kS/s (see page **43**)
- PCM data formats supported are: AES/EBU (XLR), Dual (2 wire) AES3 (XLR), Quad (4 wire) AES3 (XLR), SPDIF (Phono, Toslink and BNC) and SDIF-2 (see page **70**)
- DSD data formats supported are SDIF-2 (BNC), SDIF-3 (BNC) and DSD Quad (4 wire - XLR)

Functions

- Sample Rate and Format Conversion (page **34**)
- Multi-channel Sync capability (page **42**)
- Bit for bit multiplex/de-multiplex mode (page **39**)
- PCM to DSD, DSD to PCM
- DC Removal for DSD
- Multiple filters on many major sample rate conversions (page **45**)
- Dither – 3 types (see page **47**)
- Noise shaping – 10 different options on all major PCM sample rates (page **47**).
- Output Level control with "Maximise" (page **48**)
- Balance control (page **48**).
- Digital Silence out with digital silence in (page **50**)

Syncing

Comprehensive - can sync to Wordclock or AES reference, or signal, and sync to video option available.

[2] free if we email them, and you download from a PC COM port. Low cost if you ask us for EPROMs or other media - we charge for media and handling.

Monitoring

Bit Activity (page **56**), Stereo Output Level (page **57**), and CRC, Parity & Invalid flag errors in the input data (page **52**).

Test Generator

High quality (160 dB) signal generator with mHz resolution (page **53**). Can be dithered and/or noise shaped truncated.

Ease of Use

User programmable set-ups (page **61**)
Pre-loaded setups
Remembers last settings
Lockouts (page **64**)

About this Manual

If you have not used a ***dCS 974*** before, please read the section **Using Your dCS 974 For The First Time** on page **110**.

This manual has been arranged with the most commonly used sections placed first:

- table of contents (page **6**)
- step-by-step (page **10**) and applications guides (page **22**)
- detailed software and hardware information (page **34**)
- technical information (page **70**)
- information for first time users (page **110**)
- options, maintenance and troubleshooting (page **114**)
- index section (page **124**)

References to other sections in the text have the **Section Name**, page … with **Section Name** in bold. Sometimes, if you are reading a soft copy of the manual, section names and page numbers are hyperlinks – click on them, and you will go there.

IMPORTANT!

Important information is presented like this - ignoring this may cause you to damage the unit, or invalidate the warranty.

The manual is designed to be helpful. If there are points you feel we could cover better, or that we have missed out - please tell us.

About Sample Rates

All references to sample rates in this manual use the unit kS/s (kilo Samples per second) rather than the technically incorrect kHz.

CONTENTS

STEP-BY-STEP GUIDE

This section guides you through setting up the unit for basic operation. You may find this useful if you have not used the ***dCS 974*** for a while.

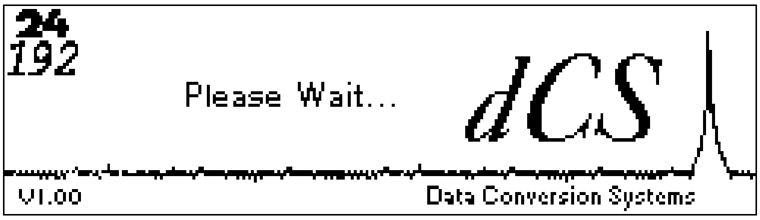
Preliminaries

The **Quick Start Guide** sheet details the menu structure and outlines the use of the front panel controls. For more information, see **Navigating through the Menu – what the On-Screen symbols mean** on page **36** and **The Software – Menu and Setups** on page **34**. We will be changing settings in either the Sample Rate Conversion menu or the Format Conversion menu. Use the rotary control to scroll up and down the screen and the **Operation** buttons to change menu levels or select items.

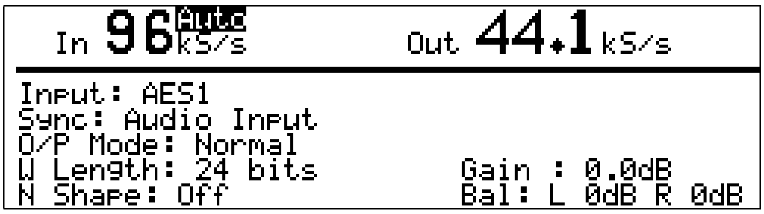
Connect up with cables designed for digital audio:

- for AES/EBU interfaces use 110Ω screened, twisted pair cables fitted with one male XLR connector and one female XLR connector.
- for DSD/SDIF or SPDIF BNC interfaces, use 75Ω coax cables fitted with BNC plugs.
- for SPDIF RCA interfaces, use 75Ω coax cables fitted with RCA Phono plugs.
- for SPDIF TOS interfaces, use Toslink fibre-optic cables.

Power up the unit and wait for about 20 seconds while it configures itself. The screen will show:



Press the **Recall** button. When the screen displays the **Recall Setup** list, press the **Recall** button again to change to the preset setup list, then press the **Enter** button. Wait while the unit loads the default setup (Store A) then displays a **Status** screen similar to this:



The **Power** and **Unlocked** indicators should be lit, the other indicators should be off.

Step 1 – Selecting an Input

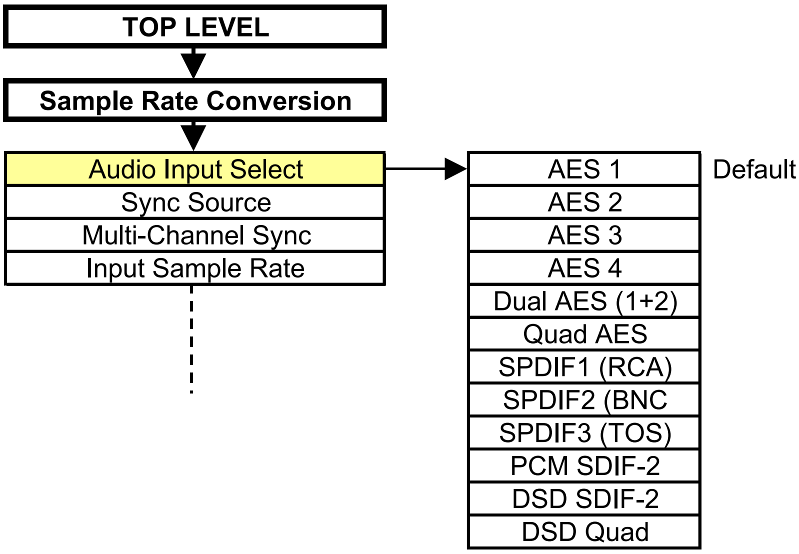


Figure 1 – Audio Input Selection

Choose one of the following five sections:

Connecting to a Single AES or SPDIF source

do this: Connect your source equipment to the matching input on the ***dCS 974*** rear panel using suitable cables. An AES3 source (XLR connector) may be connected to any of the four AES/EBU inputs.

do this: Press the → button twice to enter first the Sample Rate Conversion menu, then the Audio Input Select menu. Use the rotary control to scroll down the list until the cursor is beside your chosen input (either AES 1, AES 2, AES 3, AES 4, SPDIF 1 (RCA), SPDIF2 (BNC) or SPDIF3 (Toslink)). Press the **Set** button.

The screen will change back to the Sample Rate Conversion menu. **Proceed to Step 2.**

Connecting to a PCM SDIF-2 source

do this: Connect the SDIF-2 output on your source equipment to the upper block of DSD/SDIF connectors on the ***dCS 974*** rear panel using 3 coax cables. Connect CH1 out to **CH1 IN**, CH2 out to **CH2 IN**, CLK out to **WCLK IN**. Fit a 75Ω terminating plug to the nearby **LOOP OUT** connector.

do this: Press the → button twice to enter first the Sample Rate Conversion menu, then the Audio Input Select menu. Use the rotary control to scroll down the list until the cursor is beside PCM SDIF-2. Press the **Set** button to select it.

The screen will change back to the Sample Rate Conversion menu. **Proceed to Step 2.**

Connecting to a Dual AES Source

do this: Check that your source equipment is capable of Dual AES operation.

do this: Connect the AES 1 (or AES A) output on your source equipment to the **AES 1** input on the *dCS 974* rear panel and the AES 2 (or AES B) output to the **AES 2** input, using two XLR cables. Ensure the cables are not swapped.

do this: Press the → button twice to enter first the Sample Rate Conversion menu, then the Audio Input Select menu. Use the rotary control to scroll down the list until the cursor is beside Dual AES. Press the **Set** button to select it.

The screen will change back to the Sample Rate Conversion menu. **Proceed to Step 2.**

Connecting to a Quad AES source

do this: Check that your source equipment is capable of Quad AES operation.

do this: Connect the AES 1 output on your source equipment to the **AES 1** input on the *dCS 974* rear panel, the AES 2 output to the **AES 2** input, the AES 3 output to the **AES 3** input and the AES 4 output to the **AES 4** input, using four XLR cables. Ensure the cables are connected in the correct order.

do this: Press the → button twice to enter first the Sample Rate Conversion menu, then the Audio Input Select menu. Use the rotary control to scroll down the list until the cursor is beside Quad AES. Press the **Set** button to select it.

The screen will change back to the Sample Rate Conversion menu. **Proceed to Step 2.**

Connecting to a DSD SDIF-2 source

do this: Check that your source equipment is capable of DSD-SDIF operation.

do this: Connect the DSD SDIF-2 output on your source equipment to the upper block of DSD/SDIF connectors on the *dCS 974* rear panel using three coax cables. Connect CH1 out to **CH1 IN**, CH2 out to **CH2 IN** and CLK out to **WCLK IN.** Fit a 75Ω BNC terminating plug to the nearby **LOOP OUT** connector.

do this: Press the → button twice to enter first the Sample Rate Conversion menu, then the Audio Input Select menu. Use the rotary control to scroll down the list until the cursor is beside DSD SDIF-2. Press the **Set** button to select it.

There will be a noticeable delay while the DSD code loads, then the screen will change back to the Sample Rate Conversion menu. The *dCS 974* will automatically detect either SDIF-2 or SDIF-3. **Proceed to Step 2.**

Connecting to a DSD Quad source

do this: Check that your source equipment is capable of DSD Quad operation.

do this: Connect the AES 1 output on your source equipment to the **AES 1** input on the *dCS 974* rear panel, the AES 2 output to the **AES 2** input, the AES 3 output to the **AES 3** input and the AES 4 output to the **AES 4** input, using four XLR cables. Ensure the cables are connected in the correct order.

do this: Press the → button twice to enter first the Sample Rate Conversion menu, then the Audio Input Select menu. Use the rotary control to scroll down the list until the cursor is beside DSD Quad. Press the **Set** button to select it.

There will be a noticeable delay while the DSD code loads, then the screen will change back to the Sample Rate Conversion menu. **Proceed to Step 2.**

Step 2 – Setting the Sync Source

do this: Switch on the source equipment. If appropriate, load a disk / tape and set the machine in PLAY mode to ensure it is generating a digital audio data stream.

The ***dCS 974*** will be set to sync to the selected Audio Input and the Input Sample Rate will be Auto detected. The unit should lock and the **Unlocked** indicator should turn off. If you do not want to use an external reference clock, **proceed to Step 3**.

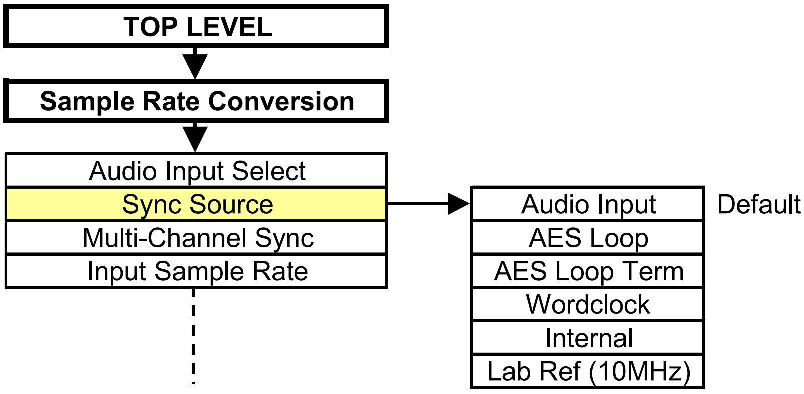


Figure 2 – Sync Source Selection

If a stable clock source is available, you can reduce jitter in your system by syncing to it. Choose one of the following two sections:

Syncing to an External Wordclock

If you want to synchronise your system to Wordclock from a Master Clock (such as the ***dCS 992***) or other stable source, do the following:

do this: Set the Master Clock sample rate to match the source (probably 44.1 or 48kS/s).

do this: Connect either a Wordclock or AES/EBU output from the Master Clock to the clock input on the source equipment and ensure it is locked.

do this: Connect another Wordclock output from the Master Clock to the **WCLK IN** connector (upper block of DSD/SDIF connectors) on the ***dCS 974*** rear panel. Fit a 75Ω BNC terminating plug to the nearby **LOOP OUT** connector.

If the source equipment uses SDIF-2 (in either PCM or DSD mode), the Wordclock feed from the Master Clock replaces the Wordclock feed from the source equipment.

do this: Scroll down the Sample Rate Conversion menu to Sync Source and press the → button. Scroll down the list to Wordclock and press **Set**.

The **Unlocked** indicator will light for a few seconds, then turn off as the unit re-locks.

do this: **Proceed to Step 3.**

Syncing to an AES/EBU Reference

If you want to synchronise your system to an AES/EBU Reference from a Master Clock (such as the ***dCS 992***) or other stable source, do the following:

do this: Set the Master Clock sample rate to match the source (probably 44.1 or 48kS/s).

do this: Connect either an AES/EBU or Wordclock output from the Master Clock to the clock input on the source equipment and ensure it is locked.

do this: Connect another AES/EBU output from the Master Clock to the **AES Ref Loop IN** connector on the ***dCS 974*** rear panel.

do this: Scroll down the Sample Rate Conversion menu to Sync Source and press the → button. Scroll down the list to AES Loop Terminated and press **Set**.

The **Unlocked** indicator will light for a few seconds, then turn off as the unit re-locks.

do this: **Proceed to Step 3.**

Step 3 - Setting a Conversion

do this: If you need bit-for-bit operation in a different output format, **proceed to the Format Conversion section**.

do this: If you want to change the sample rate or the word length or process the data in some other way, **proceed to the Sample Rate Conversion section**.

Format Conversion

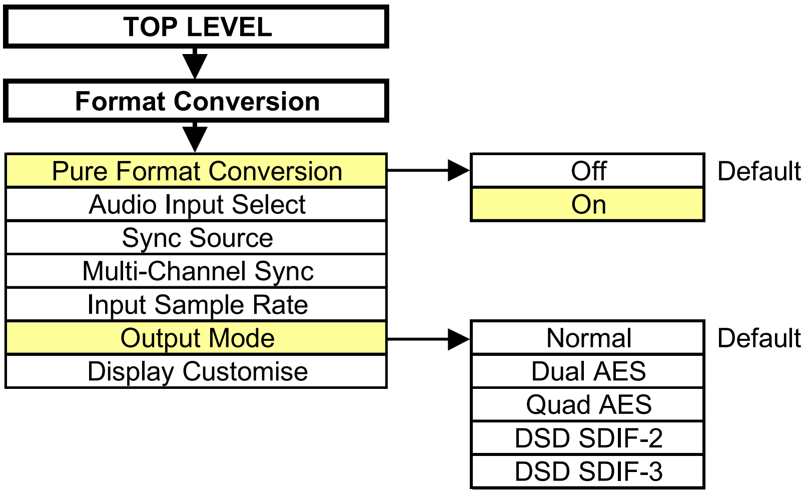


Figure 3 – Pure Format Conversion

do this: Press the ← button, scroll down to Format Conversion , press the → button and press the **Set** button. This sets Pure Format Conversion to On and disables the Sample Rate Conversion menu.

do this: If the "**Fs In not Fs Out**" information box appears on the display, press the **Set** button to make the Output Sample Rate match the Input Sample Rate.

do this: Scroll down the Format Conversion menu to Output Mode and press the → button. The cursor should be beside Normal. Choose one of the settings from the following list, scroll to it and press **Set**:

- **Normal**. The Input & Output Sample Rate must not be higher than 96kS/s. Bit-for-bit data will be available on all of the AES, SPDIF or SDIF-2 outputs.
- **Dual AES**. The Input & Output Sample Rate must be 88.2, 96, 176.4 or 192kS/s. Dual AES bit-for-bit data will be available on the **AES 1** / **AES 2** output pair and the **AES 3** / **AES 4** output pair. Do not use the other outputs.
- **Quad AES**. The Input & Output Sample Rate must be 176.4 or 192kS/s. Quad AES bit-for-bit data will be available on the **AES 1, AES 2, AES 3** and **AES 4** output group. Do not use the other outputs.
- **DSD SDIF-2**. The Audio Input Select setting must be DSD or DSD Quad. DSD SDIF-2 bit-for-bit data will be available from the **DSD/SDIF** outputs (lower block) and DSD Quad data from the **AES 1, AES 2, AES 3** and **AES 4** output group. Do not use the other outputs.
- **DSD SDIF-3**. The Audio Input Select setting must be DSD or DSD Quad. DSD SDIF-3 bit-for-bit data will be available from the **DSD/SDIF** outputs (lower block) and DSD Quad data from the **AES 1, AES 2, AES 3** and **AES 4** output group. Do not use the other outputs.

do this: **Proceed to Step 4.**

Sample Rate Conversion

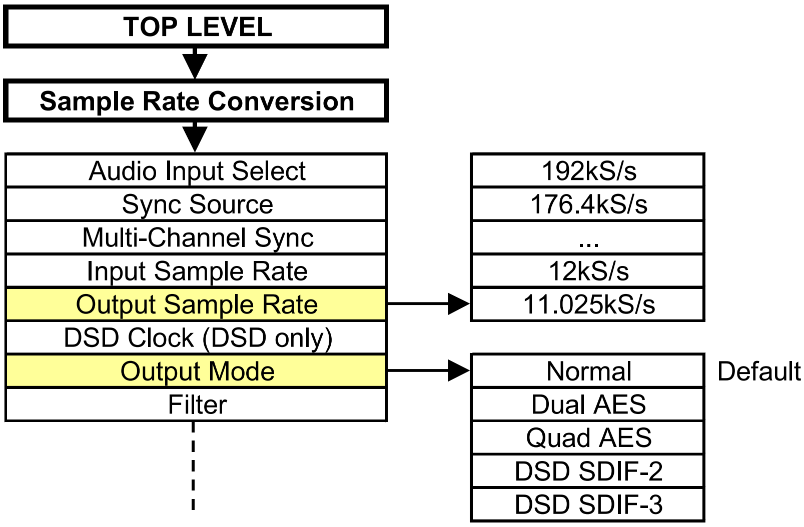


Figure 4 – Sample Rate Conversion

do this: If DSD output formats are required, **proceed to Setting the Output Mode**.

Setting the Output Sample Rate

do this: If PCM outputs are required, scroll down the Sample Rate Conversion menu to Output Sample Rate and press the → button. Scroll down the list to the required rate and press the **Set** button.

If the selected conversion can be handled in one pass, the setting will be accepted and the screen will change back to the Sample Rate Conversion menu. If not, this information box will appear on the display:

do this: Press any button to display a list of valid output sample rates. Scroll down the list to a suitable rate and press the **Set** button.

Setting the Output mode

do this: Scroll down the Sample Rate Conversion menu to Output Mode and press the → button. The cursor should be beside Normal Choose one of the settings from the following list, scroll to it and press **Set**:

- **Normal**. The Output Sample Rate must not be higher than 96kS/s. Single wire data will be available on all of the AES, SPDIF or SDIF-2 outputs.
- **Dual AES**. The Output Sample Rate must be 88.2, 96, 176.4 or 192kS/s. Dual AES data will be available on the **AES 1** / **AES 2** output pair and the **AES 3** / **AES 4** output pair. Do not use the other outputs.
- **Quad AES**. The Output Sample Rate must be 176.4 or 192kS/s. Quad AES data will be available on the **AES 1, AES 2, AES 3** and **AES 4** output group. Do not use the other outputs.
- **DSD SDIF-2**. The input format must be DSD, DSD Quad or PCM at 44.1kS/s or more. DSD SDIF-2 data will be available from the **DSD/SDIF** outputs (lower block) and DSD Quad data from the **AES 1, AES 2, AES 3** and **AES 4** output group. Do not use the other outputs.
- **DSD SDIF-3**. The input format must be DSD, DSD Quad or PCM at 44.1kS/s or more. DSD SDIF-2 data will be available from the **DSD/SDIF** outputs (lower block) and DSD Quad data from the **AES 1, AES 2, AES 3** and **AES 4** output group. Do not use the other outputs.

do this: **Proceed to Step 4.**

Step 4 – Connecting the Outputs

Choose one of the following five sections:

Connecting a Single AES or SPDIF Output

do this: If the Output Sample Rate is 88.2 or 96kS/s, check that your destination equipment is capable of double speed operation.

do this: If you have set Output Mode to Normal, connect the required single wire output on the ***dCS 974*** rear panel to the matching inputs on the destination equipment using suitable cables. Signals are available from any of the four AES/EBU outputs or the three SPDIF outputs simultaneously.

Connecting the SDIF-2 Output

do this: If the Output Sample Rate is 88.2 or 96kS/s, check that your destination equipment is capable of double speed operation.

do this: If you have set Output Mode to Normal, connect the lower block of DSD/SDIF connectors on the ***dCS 974*** rear panel to the destination equipment using 3 coax cables. Connect **CH1 OUT** to CH1 in, **CH2 OUT** to CH2 in and **WCLK OUT** to CLK in.

Connecting the Dual AES Outputs

do this: Check that your destination equipment is capable of Dual AES operation.

do this: If you have set Output Mode to Dual AES, connect the **AES 1** output on the ***dCS 974*** rear panel to the AES 1 (or AES A) input on the destination equipment and the **AES 2** output to the AES 2 (or AES B) input, using two XLR cables. Ensure the cables are not swapped. An identical Dual AES pair is available from the **AES 3** and **AES 4** outputs.

Connecting the Quad AES or DSD Quad Outputs

do this: Check that your destination equipment is capable of Quad AES or DSD Quad operation.

do this: If you have set Output Mode to Quad AES or DSD Quad, connect the **AES 1** output on the ***dCS 974*** rear panel to the AES 1 input on the destination equipment, the **AES 2** output to the AES 2 input, the **AES 3** output to the AES 3 input and the **AES 4** output to the AES 4 input, using four XLR cables. Ensure the cables are not swapped.

Connecting the DSD SDIF-2 or DSD SDIF-3 Output

do this: Check that your destination equipment is capable of DSD operation.

do this: If you have set Output Mode to DSD SDIF-2 or DSD SDIF-3, connect the lower block of DSD/SDIF connectors on the ***dCS 974*** rear panel to the destination equipment using three coax cables. Connect **CH1 OUT** to CH1 in, **CH2 OUT** to CH2 in and **WCLK OUT** to CLK in.

Note that the default setting for the DSD output clock is 44.1kS/s Wordclock (rather than Bit clock at 2.82MS/s).

do this: **Proceed to Step 5.**

Step 5 – Reducing the Output Wordlength

If you are using Pure Format Conversion or DSD output modes, the Output Wordlength cannot be changed. **Proceed to Other Settings**.

If you are performing a sample rate conversion with PCM outputs, the destination equipment can handle 24 bit data and you do not want to reduce the wordlength then leave the Output Wordlength set to the default of 24 bits and set Dither to Off. **Proceed to Other Settings**.

The ***dCS 974*** generates 24 bit data, regardless of the input word length. If the destination equipment cannot handle 24 bit data, the Output Wordlength MUST be set to match. Noise Shaping and/or Dither MUST be applied to smooth the transitions. If the extra bits are just ignored, the audio outputs may sound grainy and unpleasant low effects will result. For more information, see **Word Length Reduction** on page **104**.

do this: Check the maximum input wordlength specification in the manual for the destination equipment. You must set the ***dCS 974*** to match this.

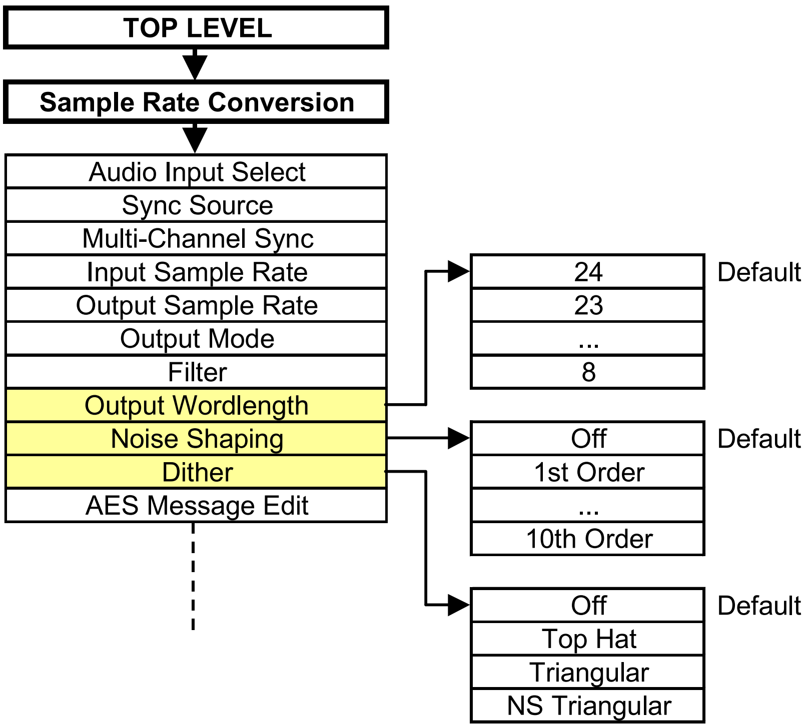


Figure 5 – Setting Wordlength, Noise Shaping and Dither

do this: Scroll down the Sample Rate Conversion menu to Output Wordlength and press the → button. Scroll down the list to the required number of output bits and press **Set**.

do this: Scroll down the Sample Rate Conversion menu to Noise Shaping and press the → button. From the list below, choose a suitable setting to match the Output Wordlength:

- for 22 or 23 bits, scroll to 2nd order and press **Set**.
- for 20 or 21 bits, scroll to 3rd order and press **Set**.
- for 16, 17, 18 or 19 bits, scroll to 9th order and press **Set**.

do this: For 16 or 17 bits, scroll down the Sample Rate Conversion menu to Dither and press the → button. Scroll down to NS Triangular and press **Set**.

A wide variety of Noise Shaping and Dither setting combinations are possible. For more information, see **Word Length Reduction** on page **104**.

do this: **Proceed to Other Settings.**

Other Settings

The basic set-up procedure is complete. The Sample Rate Conversion menu contains several other menu pages. For more information, see **Sample Rate Conversion / Format Conversion Submenu**, starting on page **41**.

Typical Applications

Converting a 24/96 recording to CD format

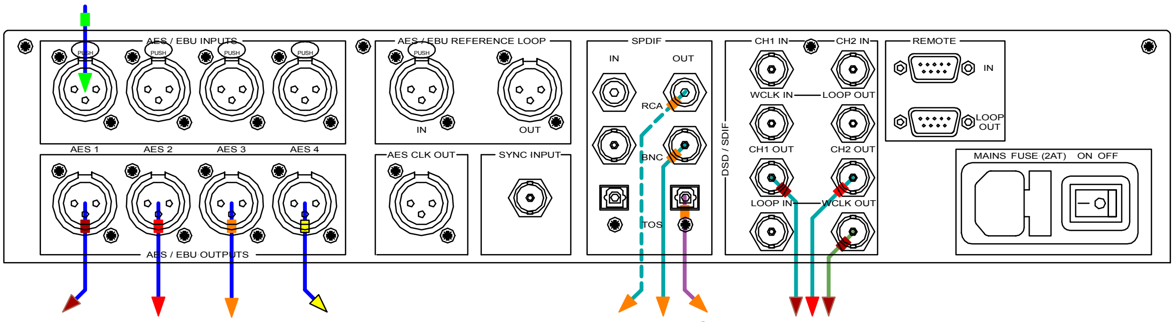


Figure 6 – Double Speed 24/96 to CD format

The ***dCS 974*** converts a double speed 96 kS/s 24 bit AES input to a 44.1 kS/s 16 bit signal available from all of the 4 AES, 2 electrical SPDIF, optical SPDIF or SDIF-2 outputs. The conversion set up uses Filter 2 (there is a choice of 4). The SonicStudio™ uses 24/96 as double speed AES at the time of writing.

do this: Connect Sonic Solutions SonicStudio™ workstation AES output to **AES 1** input on the ***dCS 974.***

do this: Load the setup from Store K, or use the settings below.

do this: Output from any AES output or any of the SPDIF outputs or, using two data cables and one clock, via the SDIF-2 outputs.

Sample Rate Conversion settings:

Sample Rate Conversion:	On
Audio Input Select:	AES 1
Sync Source:	Audio Input
Input Sample Rate:	Auto (96 kS/s)
Output Sample Rate:	44.1 kS/s
Output Mode:	Normal
Filter:	Filter 2
Output Wordlength:	16
Noise Shaping:	9th Order
Dither:	Off
Detect Silence:	On
AES Message Edit:	Professional Off, Non-Audio Off Mode: Stereophonic
SPDIF Message Edit:	Professional Off, Non-Audio Off, Copy Permit On Format: Compact Disc
Gain:	-0.1dB

Demultiplexing a 24/96 Dual AES recording (Bit for Bit)

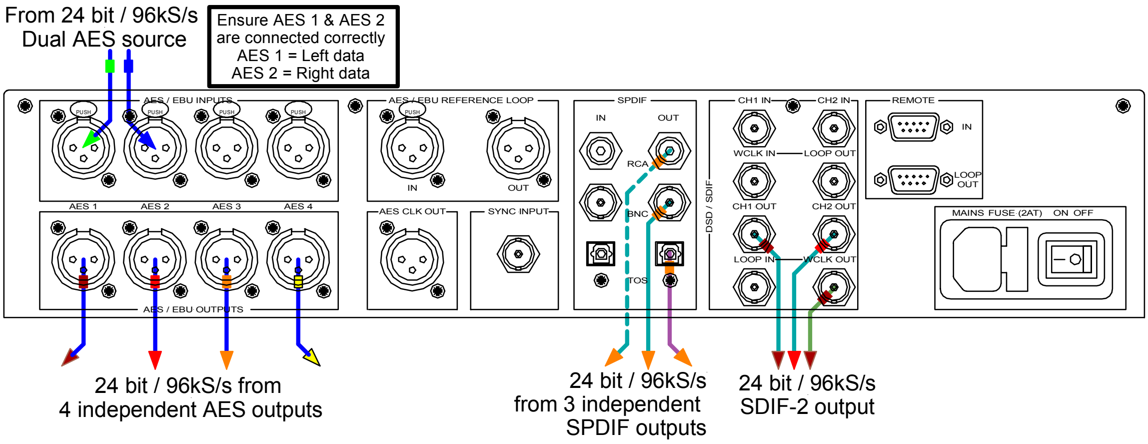


Figure 7 – Converting dual AES 24/96 to a single wire double speed one

The ***dCS 974*** converts a 96 kS/s 24 bit Dual AES input to a double speed 96 kS/s 24 bit signal available from any or all of the 4 AES, 2 electrical SPDIF, optical SPDIF or SDIF-2 outputs. When using the Format Conversion menu, the operation is bit for bit on the audio data (messages can be edited).

do this: Connect Nagra-D, Genex G-8000, SADIE, Lake DSP, etc dual AES source to inputs **AES 1** and **AES 2** on the ***dCS 974***.

do this: Load the setup from Store J and set Pure Format Conversion to On, or use settings below.

do this: Output from any AES output or any of the SPDIF outputs or, using two data cables and one clock, via the SDIF-2 outputs. The outputs will all be at "double" speed.

Format Conversion settings:

Pure Format Conversion:	On
Audio Input Select:	Dual AES
Sync Source:	Audio Input
Input Sample Rate:	Auto (96 kS/s)
Output Sample Rate:	96 kS/s
Output Mode:	Normal
AES Message Edit:	Professional On, Non-Audio Off Mode: Stereophonic
SPDIF Message Edit:	Professional On, Non-Audio Off, Copy Permit On Format: 2-Ch Gen Format

Upsampling a CD

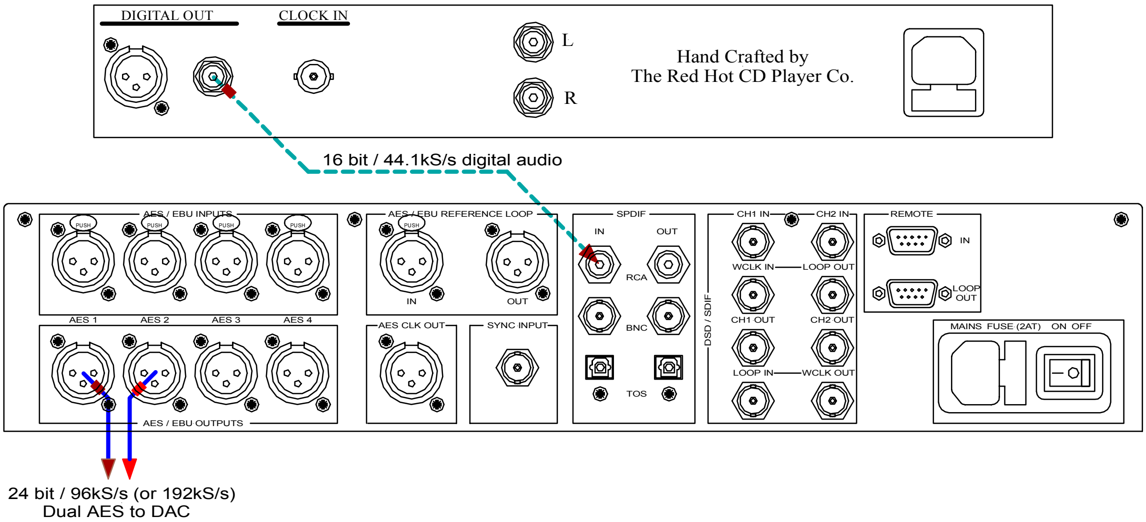


Figure 8 – Upsampling a CD to 24/96 (or 24/192)

The ***dCS 974*** converts a 44.1 kS/s 16 bit SPDIF input to a Dual AES 96 kS/s 24 bit signal available from **AES 1** & **AES 2** outputs. There is no information added in the process, although you may wish to check theory here.

do this: Connect a source of CD material to the **RCA** input on the ***dCS 974.***

do this: Enter the Sample Rate Conversion settings below.

do this: Connect Dual AES data from both **AES 1** and **AES 2** outputs to a DAC of your choice.

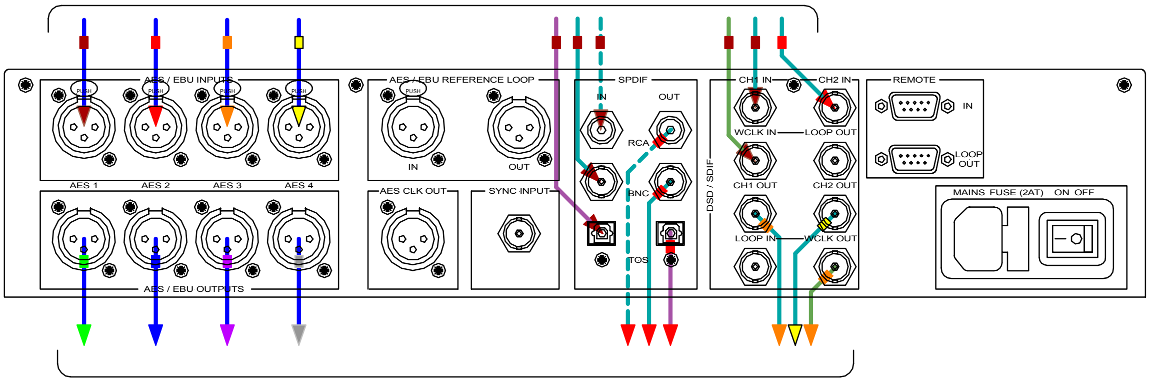
do this: If your DAC is a ***dCS Elgar***, ***dCS Delius*** or ***dCS 954***, you can set the Output Sample Rate to 192kS/s instead.

Sample Rate Conversion settings:

Sample Rate Conversion:	On
Audio Input Select:	SPDIF 1
Sync Source:	Audio Input
Input Sample Rate:	Auto (44.1 kS/s)
Output Sample Rate:	96 kS/s
Output Mode:	Dual AES
Filter:	Filter 2
Output Wordlength:	24
Noise Shaping:	Off
Dither:	Off
Detect Silence:	Off
AES Message Edit:	Professional On, Non-Audio Off, Mode: Stereophonic
SPDIF Message Edit:	Professional On, Non-Audio Off, Copy Permit On Format: 2-Ch Gen Format

General Sample Rate Conversion and Distribution

From ANY format: AES, SPDIF, Toslink, SDIF-2 at ANY sample rate:
11.025k, 12k, 16k, 22.05k, 24k, 32k, 44.1k, 48k, 88.2k or 96kS/s
or
Dual AES at 88.2k, 96k, 176.4k or 192kS/s
or
Quad AES at 176.4 or 192kS/s
or
DSD SDIF-2, DSD SDIF-3 or DSD Quad



To ALL formats: AES, SPDIF, Toslink & SDIF-2 at ANY sample rate:
11.025k, 12k, 16k, 22.05k, 24k, 32k, 44.1k, 48k, 88.2k or 96kS/s
or
Dual AES at 88.2k, 96k, 176.4k or 192kS/s
or
Quad AES at 176.4 or 192kS/s
or
DSD SDIF-2, DSD SDIF-3 or DSD Quad

Figure 9 – General Sample Rate Conversion

The ***dCS 974*** converts any one of 12 sample rates in any format to any of the 12 sample rates in ALL formats: 4 AES, 2 electrical SPDIF, optical SPDIF or SDIF-2 outputs. Most frequency combinations (including all the primary ones) are accommodated in one pass - see **Table 2** on page **44**. For best results, slave the unit to the audio input, **AES Ref Loop In** or the SDIF-2 **WCLK IN**.

do this: Connect any digital audio source of any word length between 8 and 24 bits, using single AES or dual AES or quad AES to the AES inputs, or using single or double speed SDIF-2 to the SDIF-2 inputs, or using single wire SPDIF to the appropriate SPDIF input.

do this: Select Audio Input accordingly. Output via any of the outputs.

Sample Rate Conversion settings:

Sample Rate Conversion:	On
Audio Input Select:	Any, including Dual AES if the Input Sample Rate is 88.2kS/s or more and Quad AES if the Input Sample Rate is 176.4 kS/s[3] or 192kS/s.
Output Mode:	Normal or Dual AES may be selected if the Output Sample Rate is 88.2 kS/s or 96kS/s. Dual AES or Quad AES must be used for 192 kS/s or 176.4 kS/s.
Other settings:	Any.

[3] It has to be Dual AES or Quad AES for 192 kS/s or 176.4 kS/s

PCM to DSD

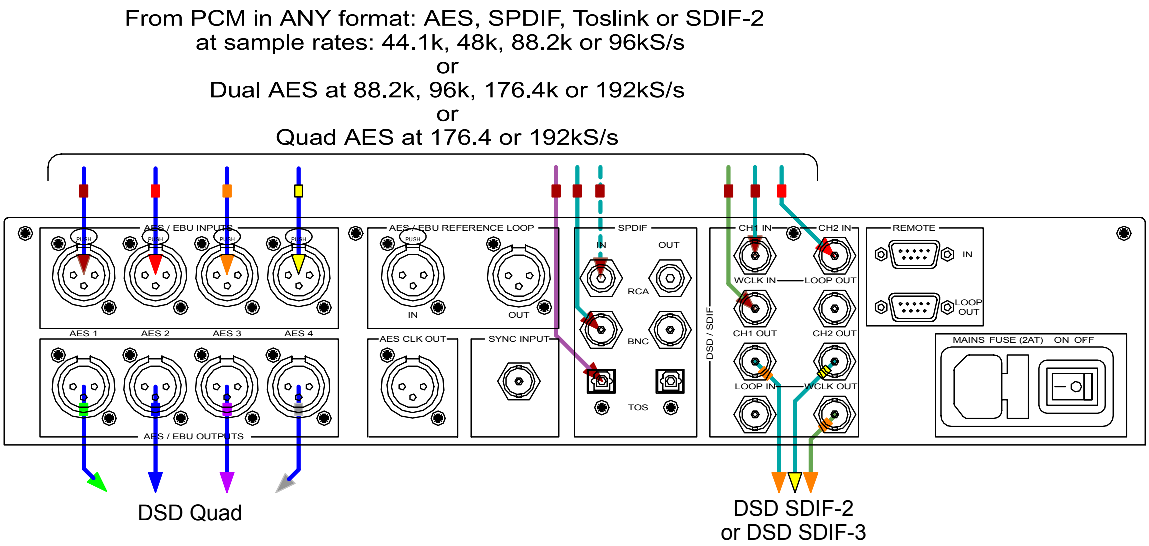


Figure 10 – PCM to DSD conversion

The ***dCS 974*** converts PCM to DSD (including 176.4 kS/s to DSD) using the arrangement given below:

do this: Connect any digital audio source of any wordlength between 8 and 24 bits, using single, Dual or Quad AES to the AES inputs, or using single or double speed SDIF-2 to the SDIF-2 inputs, or using single wire SPDIF to the appropriate SPDIF input, and select the input accordingly.

do this: Set Output Mode to DSD SDIF-2 or DSD SDIF-3. Choose a Filter if you wish.

do this: Output SDIF-2 via the **DSD/SDIF** connectors, two data cables and one word clock or SDIF-3 with just two data cables.

do this: Alternatively, take the DSD Quad output from the **AES 1**, **2**, **3** and **4** outputs.

Sample Rate Conversion settings:

Sample Rate Conversion:	On
Audio Input Select:	Any, even including Dual AES or Quad AES at higher Input Sample Rates
Sync Source:	Any
Input Sample Rate:	Any
Output Sample Rate:	n/a
Output Mode:	DSD
Filter:	Any
Output Wordlength:	n/a
Noise Shaping:	n/a
Dither:	n/a
Detect Silence:	n/a

As an example, load Store H to take an input from **AES1** and convert it to DSD SDIF-2 or DSD Quad format.

Using a Master Clock

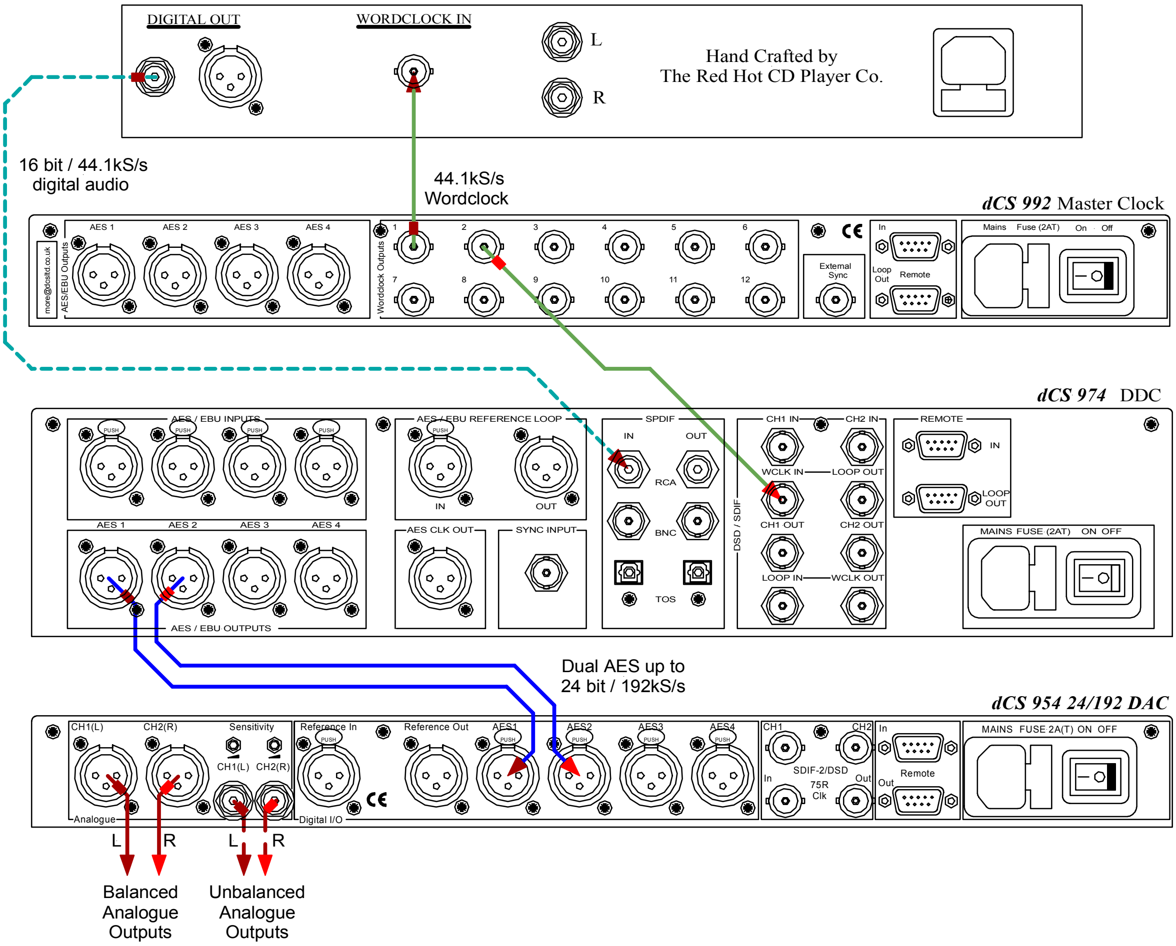


Figure 11 – Using a Master Clock

If a Master Clock such as a ***dCS 992*** is available and the driving source is locked to it, the ***dCS 974*** may be locked to it. Either:

do this: connect the Master Clock to the **AES Ref Loop In** and select AES Loop Term as the Sync Source,

or:

do this: connect the Master Clock to the SDIF-2 **WCLK IN** and select Wordclock as the Sync Source.

IMPORTANT! *Since the Output Sample Rate of the **dCS 974** is different to the Master Clock rate, the DAC would be unable to lock to the data from the **dCS 974** if it were sync'ed to the Master Clock. So, slave the DAC to the **dCS 974, NOT** to the Master Clock.*

Converting Quad AES to CD Format

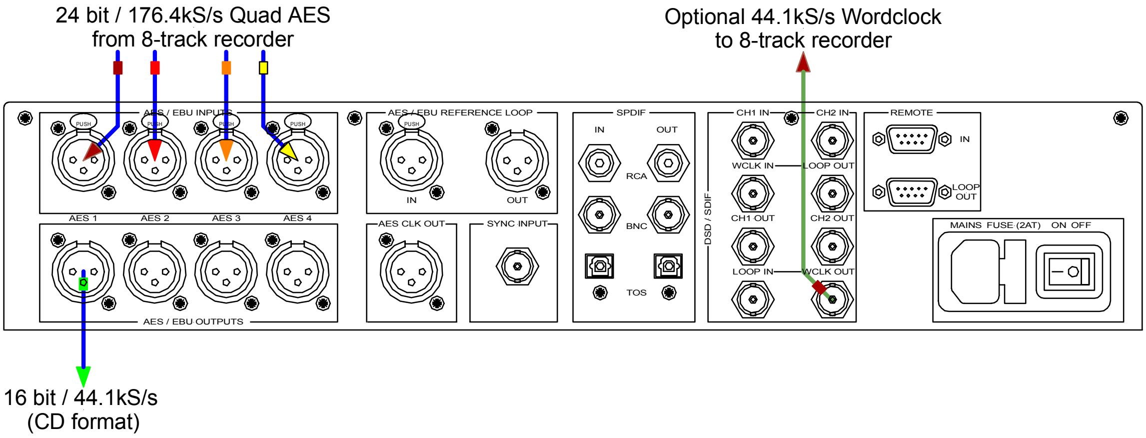


Figure 12 – Converting Quad AES to CD format

You can archive in 24 bit / 176.4kS/s or 192kS/s Quad AES format using a standard 8-track digital recorder, then convert to other formats such as Red Book CD.

Sample Rate Conversion settings:

Sample Rate Conversion:	On
Audio Input Select:	Quad AES
Sync Source:	Audio Input
Input Sample Rate:	Auto
Output Sample Rate:	44.1kS/s
Output Mode:	Normal
Filter:	Any
Output Wordlength:	16
Noise Shaping:	9th Order
Dither:	NS Triangular
Detect Silence:	On

Provided the Output Sample Rate is set to one quarter of the Input Sample Rate (i.e 176.4kS/s ⇒ 44.1kS/s or 192kS/s ⇒ 48kS/s), you can reduce jitter by using the ***dCS 974*** as the master clock and locking the recorder to it. Change Sync Source to Internal, connect **WCLK OUT** (lower block) to the recorder's Wordclock input and set the recorder to slave.

Archive at 176.4kS/s for audio-based material, archive at 192kS/s for video-based material. Ensure the sample rates match the source material.

Multi-channel Sample Rate Conversion – bit aligned sources

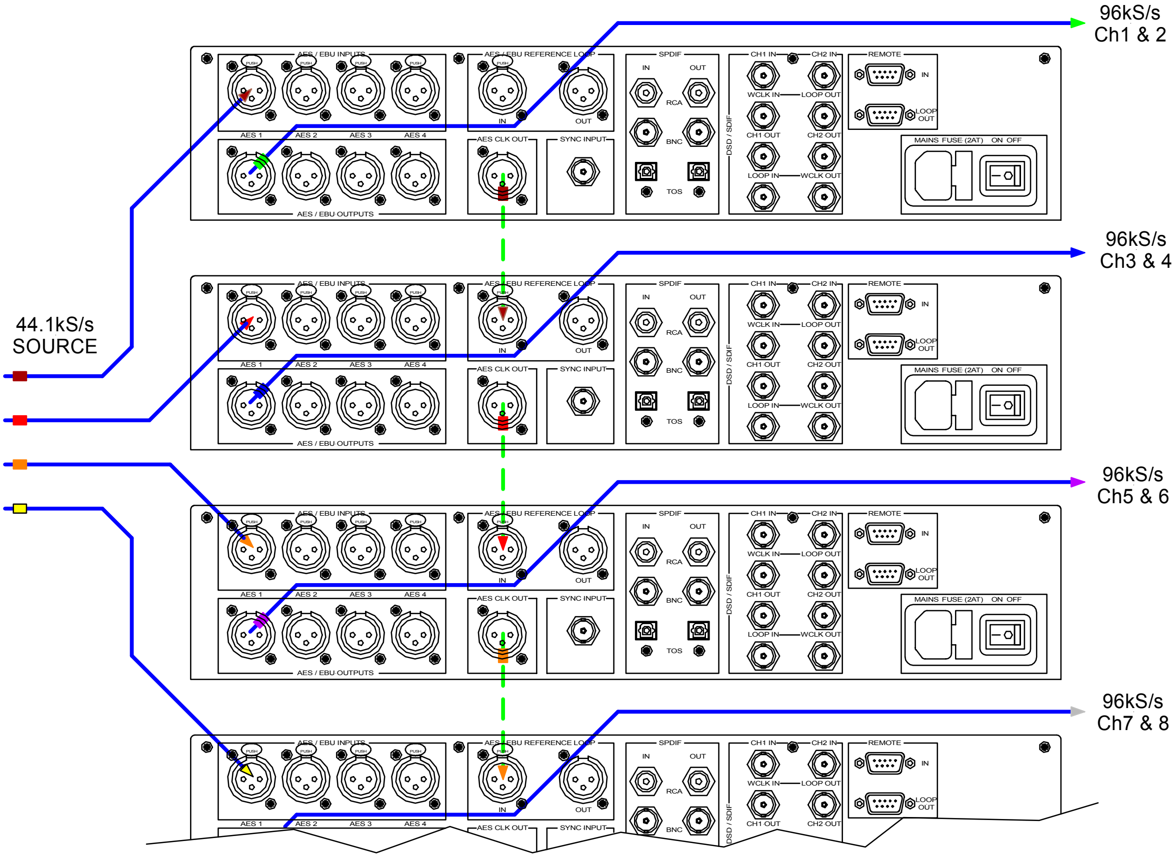


Figure 13 – Multi-channel Sample Rate Conversion with bit aligned source

If the source data is bit aligned (for example, from a multitrack recorder) up to four or five ***dCS 974***'s may be set to operate synchronously using the sync link connections shown above, from one unit's **AES CLK Out** to the next unit's **AES Ref Loop In**. For all units, set the Multiple Channel Sync option to On. The top unit will set up as a master, the other ones will set up as slaves. Make sure that all the other settings are the same on each unit.

See the section **Multiple Channel Sync'ing**, page **81** for how aligned is bit aligned.

These are set up, for PCM in to DSD out, in Store E, and for PCM in to 96kS/s out in Store F.

Multi-channel Sample Rate Conversion – Using a Master Clock

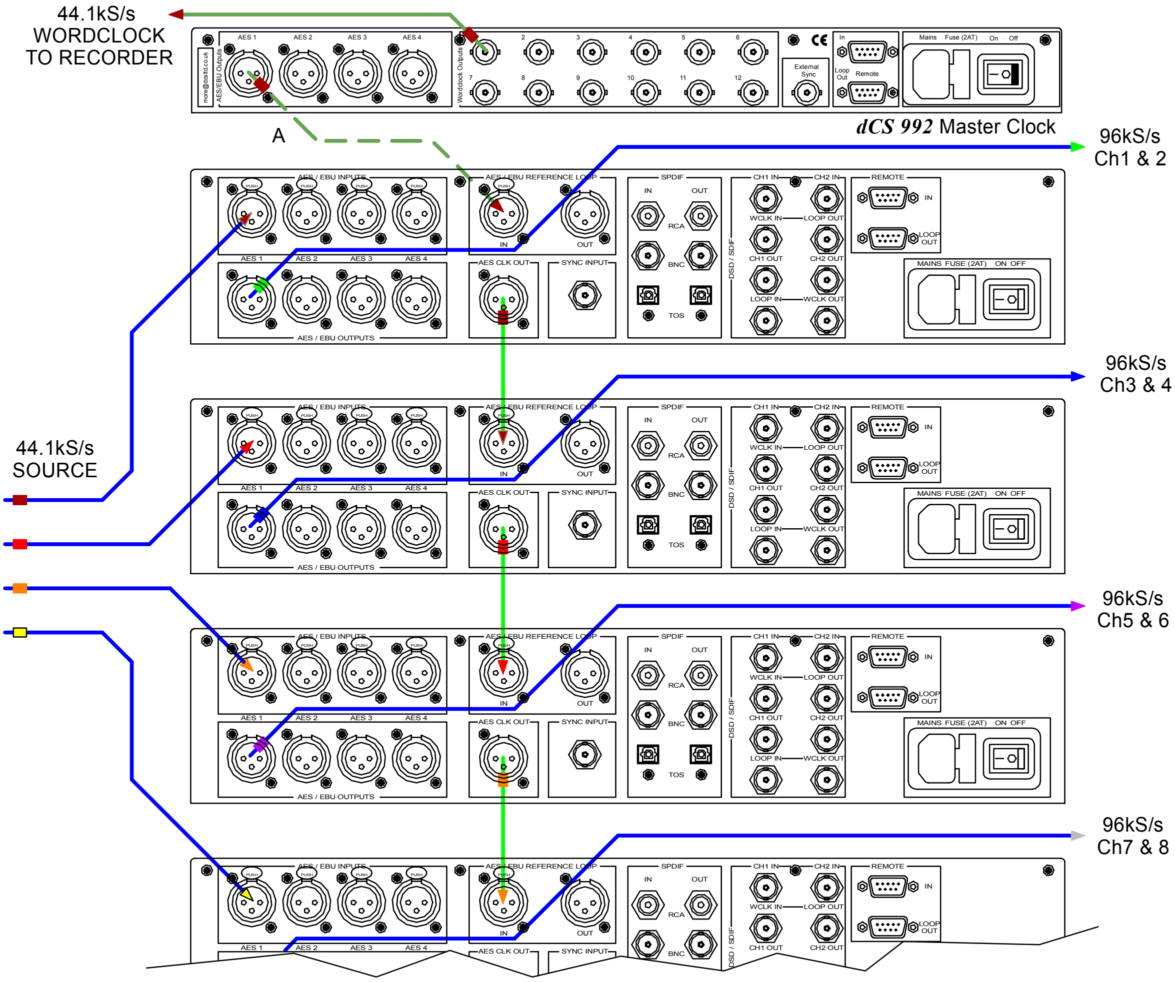


Figure 14 – Multi-channel Sample Rate Conversion with Master Clock

You can use a ***dCS 992*** Master Clock[4] to sync up the source and be the master. The sync link cabling starts from the master clock (cable "A") and then carries on down units as shown. Use of the master clock allows more units to be used together – 24 channels worth or more. For all ***dCS 974*** units, set the Multiple Channel Sync option to On. They will all set up as slaves. Make sure that all the other settings are the same on each unit.

4 version 2.0 or higher software

Multi-channel Sample Rate Conversion – with more alignment tolerance

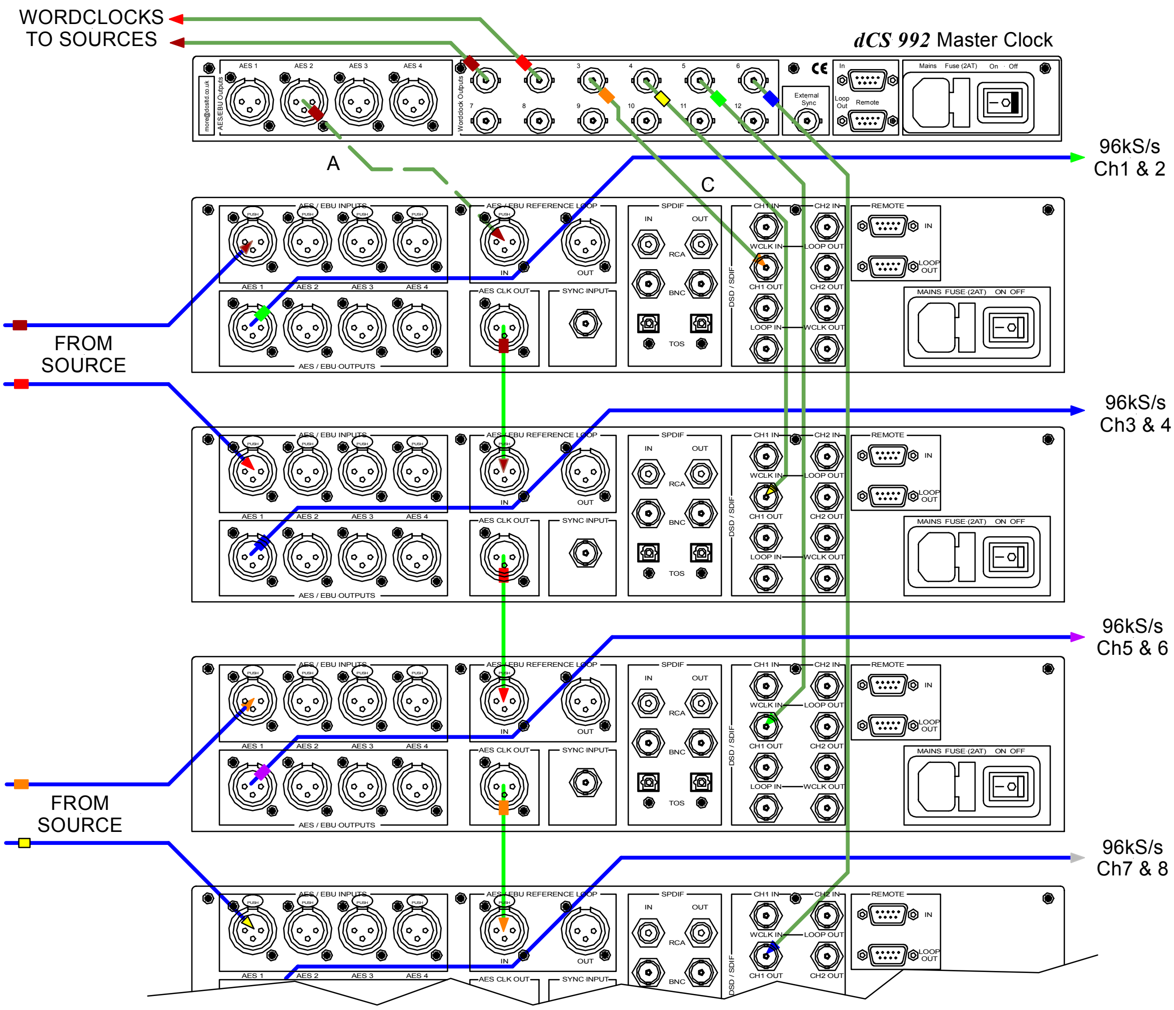


Figure 15 – Multi-channel Sample Rate Conversion with more alignment tolerance

A ***dCS 992*** Master Clock[5] can also be used to sync the source and the ***dCS 974*** units as above. In addition to the sync link cabling (starting with "A"), additional clocking (cables "C") allow the ***dCS 974***'s to extract clocks from the "C" cables and extract the data from the signal input cables. The master clock outputs different frequencies on the different cabling types. This allows considerable bit alignment error, as might occur if some tracks are stored on one machine and some on another. For all ***dCS 974*** units, set the Sync Source to Wordclock and the Multiple Channel Sync option to On. They will all set up as slaves. Make sure that all the other settings are the same on each unit. Note that cable 'A' must NOT be driven from the ***dCS 992***'s AES1 output.

5 version 2.0 or higher software

Multi-channel Sample Rate Conversion – with multiple sample rates out

A ***dCS 992*** Master Clock[6] can also be used to sync up the source and more than one set of units, all at different sample rates, and be the master. DVD production may require different sample rates for different channels – this set up will give real time 48 kS/s rear channels and 96 kS/s front channels from a multi-channel 44.1kS/s source. See **Figure 16** overleaf. For all ***dCS 974*** units, set the Multiple Channel Sync option to On. They will all set up as slaves. Make sure that all the other settings are the same on each unit in a group.

[6] version 2.0 or higher software

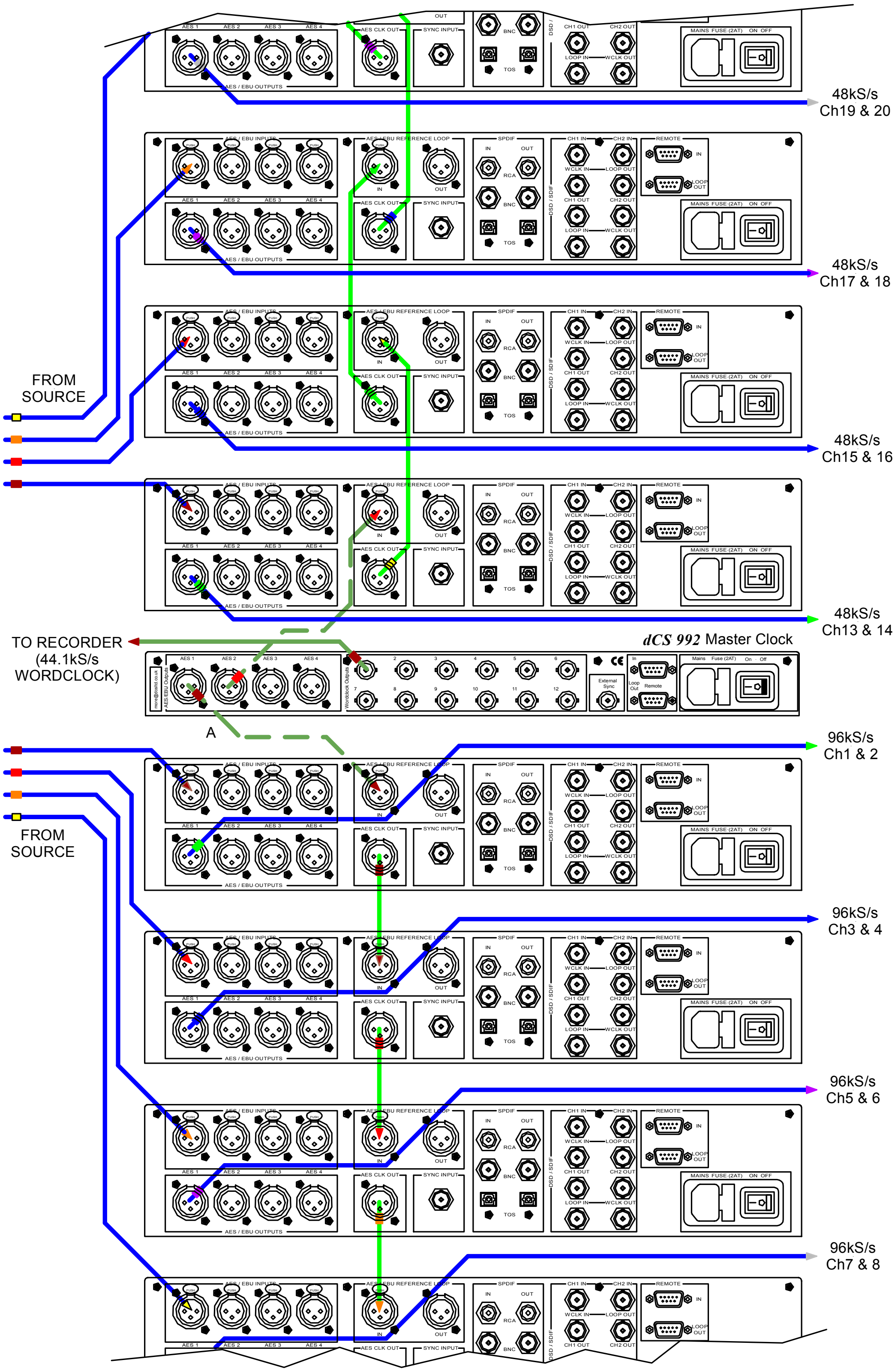
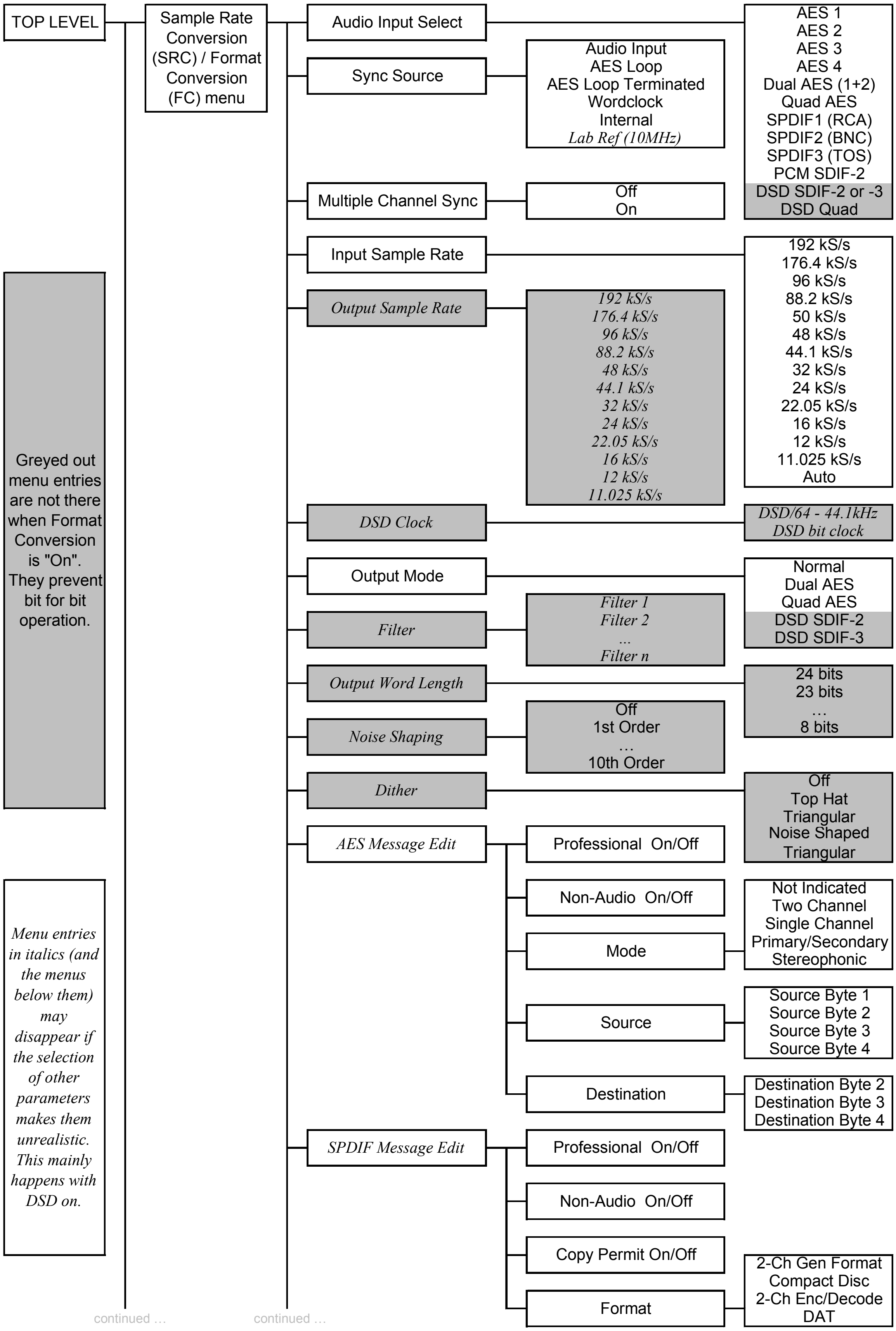


Figure 16 – Multi-channel Sample Rate Conversion with multiple sample rates out

THE SOFTWARE – MENU AND SETUPS



Contact *dCS* on + 44 1799 531 999
(inside the UK replace + 44 with 0)
email to: more@dcsltd.co.uk
web site: www.dcsltd.co.uk

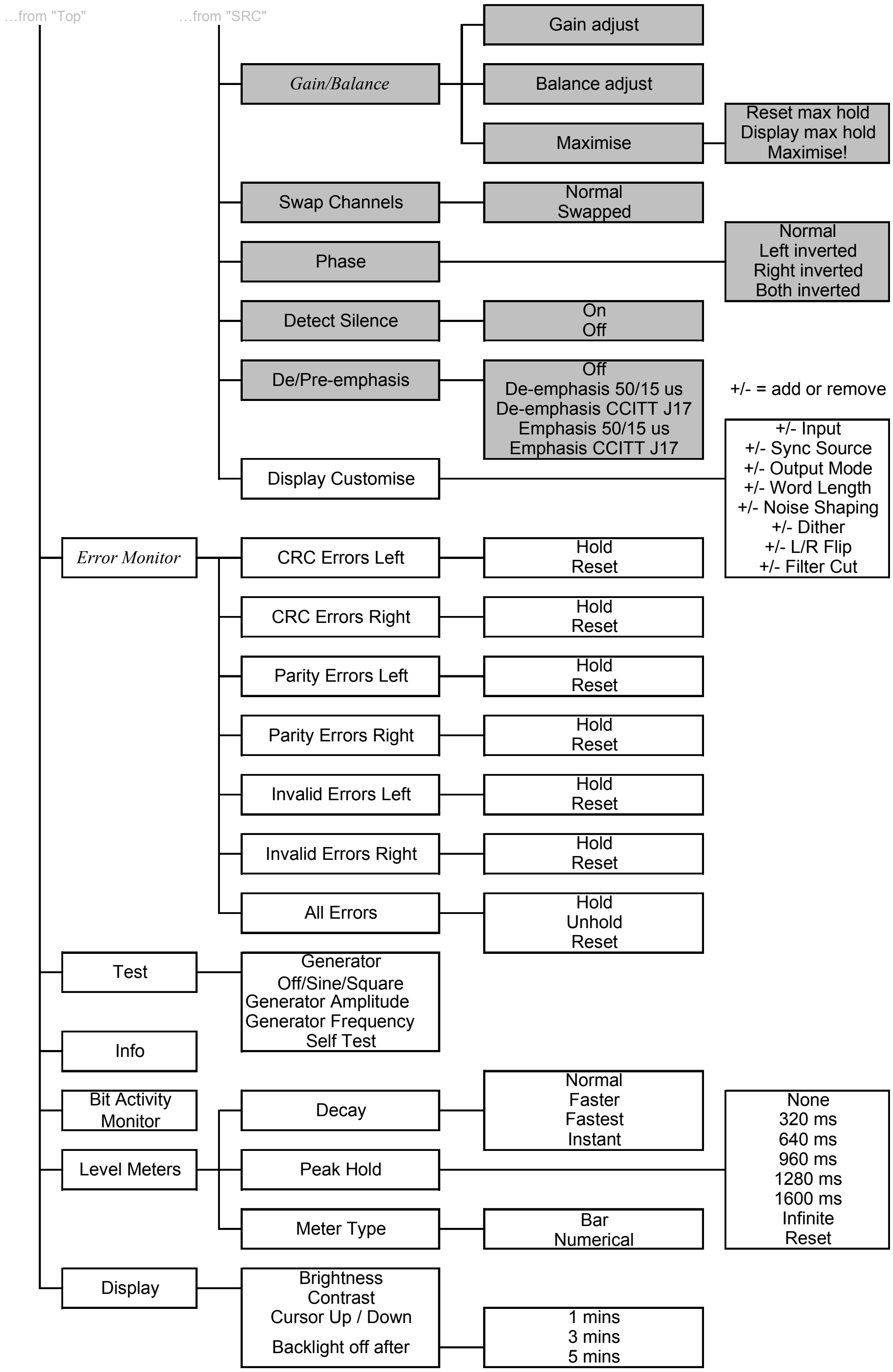


Table 1 – Menu Tree

Navigating through the Menu – what the On-Screen symbols mean

Once the unit has powered up and the **Status** screen is displayed, you can start navigating through the menu. This section explains how the keys operate, and what the various on-screen indications mean. To start, pressing either the ← or → button brings up the top level of the Menu:

```
▶Sample Rate Conversion
 Format Conversion
 Error Monitor
 Test
 Info
 Bit Activity Monitor
 Level Meters
 Display
```

The ▶ symbol is the cursor and an indication that there are sub-menus available below this level. If there are no more submenus, it changes to ■. Turn the rotary control clockwise and the cursor will move down the list. Turn the rotary control counter clockwise and the cursor will move back up. To access the Sample Rate Conversion menu, set the cursor adjacent to Sample Rate Conversion and press the → button. The screen will change to:

```
Sample Rate Conversion
   ▶Audio Input Select
    Sync Source
    Multiple Channel Sync
    Input Sample Rate
    Output Sample Rate
    Output Mode
    Filter                        ↓
```

The cursor shape is still ▶ because the option selected has a lower level. The ↓ symbol in the lower right hand corner indicates that there are more options available than can be displayed. Use the rotary control to move the cursor down the list. When the cursor reaches the bottom of the screen, the list will scroll upwards and a ↑ appear in the upper right hand corner to indicate that there are more options above. When the bottom of the list is reached, the ↓ symbol disappears.

The screen displays:

```
Sample Rate Conversion
    SPDIF Message Edit            ↑
    Gain/Balance
    Swap Channels
    Phase
    Detect Silence
    De/Pre-Emphasis
   ▶Display Customise
```

Move the cursor back up to Balance/Gain and press the → button to display the bottom level:

```
Sample Rate Conversion
   Gain/Balance
      ■Gain : -6.0dB
       Bal: L0.0dB R0.0dB
       Maximise
```

The cursor has changed to ■ to indicate that the bottom level has been reached. Turning the rotary control will move the cursor up and down the list. Now press the **Set** button to accept Gain and the cursor changes to ⁘ to indicate that this parameter (Gain) may now be adjusted using the rotary control. Turn the control either way and the Gain changes in 0.1dB steps. When the required setting is reached, press **Set** and the cursor will change back to ■.

Pressing ← when the cursor is ▶ or ■ returns to the previous menu level.

Pressing **Set** when the cursor is ■ and the option cannot be adjusted (e.g. AES 1 in the Audio Input Select menu) selects that option and returns to the previous menu level.

From the **Status** screen, pressing any of the four **Operation** buttons displays the last menu level used.

Pressing the **Status** button when the cursor is ▶ or ■ displays the **Status** screen for the selected menu.

Top Menu

The top menu contains the major entries: - Sample Rate Conversion, Format Conversion, Error Monitoring, Test, Info, Bit Activity Monitors, Level Meters and Display – as follows:

```
▶Sample Rate Conversion
 Format Conversion
 Error Monitor
 Test
 Info
 Bit Activity Monitor
 Level Meters
 Display
```

In many cases, the settings in one menu do not affect those in others. In other cases they do, and where this occurs the menu will automatically adjust to allow only valid options.

The unit can operate in one of two main modes – Sample Rate Converter (SRC) mode and Format Converter (FC) mode. In SRC mode, the unit performs DSP on the signal and gain is always reduced by 0.01dB. This is because many of the other operations, such as dithering or noise shaping, add a small amplitude signal, so we reduce the amplitude a small amount to prevent spurious clips.

It also means that for simple operations such as Dual AES in to double speed AES out, bits in will not be the same as bits out. Because in some cases this is important, the unit can also operate in FC mode. FC mode has no (0dB) signal drop, and will not allow operations that can cause audio data bit changes – but it does allow bit for bit copies of signals to be made in different digital formats. It also allows message editing, but many of the other options available in SRC mode are removed from the menus while FC mode is active.

SRC mode offers all the options of FC mode – FC mode just turns off the options that affect bit for bit performance. There are no options that are just available in FC mode.

Sample Rate Conversion

The default setting is Sample Rate Conversion (SRC) mode on, Format Conversion (FC) mode Off. Pressing → will open the Sample Rate Conversion menu.

```
Sample Rate Conversion
    ▶Audio Input Select
     Sync Source
     Multiple Channel Sync
     Input Sample Rate
     Output Sample Rate
     Output Mode
     Filter                           ↓
```

Format Conversion

The default setting is SRC mode on, FC mode Off. To turn FC mode on, select Format Conversion and press →. The menu below will appear:

```
Format Conversion
   ■Pure Format Conversion: Off
    Audio Input Select
    Sync Source
    Multiple Channel Sync
    Input Sample Rate
    Output Mode
    AES Message Edit                  ↓
```

Pressing **Set** will toggle the state from Off to On. The unit will check to see if the Output Sample Rate is the same as the Input Sample Rate, and if it is not it asks you to press **Set** to change the Output Sample Rate, or any other button to cancel the attempt to turn FC mode on.

```
Format Conversion
   ■Pure Format Conversion: On
    Audio Input Select
    Sync Source
    Multiple Channel Sync
    Input Sample Rate
    Output Mode
    AES Message Edit                  ↓
```

On returning to the main menu, Sample Rate Conversion is shown as disabled:

```
■Sample Rate Conversion (disabled)
 Format Conversion
 Error Monitor
 Test
 Info
 Bit Activity Monitor
 Level Meters
 Display
```

To re-enter SRC mode, you have to set Pure Format Conversion to Off.

Error Monitoring

Error monitoring on the input signal can be implemented for CRC, Parity and Valid bits in the AES3 and SPDIF message streams. This menu covers resetting the monitors, and is only available for PCM inputs – the entry disappears for DSD inputs.

Test

The unit can be used as a very high purity signal generator, as well as performing self test functions, via this menu.

Info

This menu displays information about the unit, for support purposes

Bit Activity Monitors

This menu turns on bit activity monitors on the selected input signal, for use on the audio data – it can find out how many active bits you have coming in, or identify stuck bits in other equipment. This only works for PCM inputs – the entry disappears for DSD inputs. This feature does **NOT** monitor the units outputs.

Level Meters

Controls level metering of the output data. The level meters may be used to monitor DSD signals by setting a DSD to PCM conversion. Even though the conversion may not be used, the level metering will effectively meter the DSD input.

Display

Controls the Brightness, Contrast, Cursor direction and Backlight timeout behaviour of the display.

Sample Rate Conversion / Format Conversion Submenu

The Sample Rate Conversion menu is as follows:

```
Sample Rate Conversion
   ▶Audio Input Select
    Sync Source
    Multiple Channel Sync
    Input Sample Rate
    Output Sample Rate
    Output Mode
    Filter                            ↓
```

Audio Input Select

Select the required input from the list and press **Set**. The Dual AES option requires that the Input Sample Rate be set to 192 kS/s, 176.4 kS/s, 96 kS/s or 88.2 kS/s. The Quad AES option requires that the Input Sample Rate be set to 192 kS/s or 176.4 kS/s. The DSD setting automatically identifies DSD SDIF-2 or DSD SDIF-3 data on the DSD/SDIF input.

Sync Source

Select a source to synchronise to from the list and press **Set**. The available sync sources are:

Audio Input	Syncs to the selected input source – this is the normal setting.
AES Loop	Syncs to the **AES Ref Loop In**, leaving the reference unterminated to allow daisy-chain connection through **Loop Out** to another device.
AES Loop Term	Syncs to the **AES Ref Loop In** and terminates the reference source.
Wordclock	Syncs to **DSD/SDIF WCLK IN** (in the upper group of connectors).
Internal	Syncs to the ***dCS 974***'s internal clock.
Lab Ref	Syncs to a 10 MHz signal into **DSD/SDIF CLK IN** (in the upper group of connectors).

For best results, synchronise to the Audio Input or a master clock on one of the clock/reference inputs that is also driving the source. For DSD/SDIF inputs, you can still sync to Audio Input as the unit will automatically sync to **DSD/SDIF WCLK IN**. For DSD in SDIF-2 or SDIF-3 mode, a Wordclock must be used – contact ***dCS*** if you need a Bit Clock.

The Lab Ref setting allows use of a GPS reference - if you use this, make sure other parts of your system are also GPS sync'd.

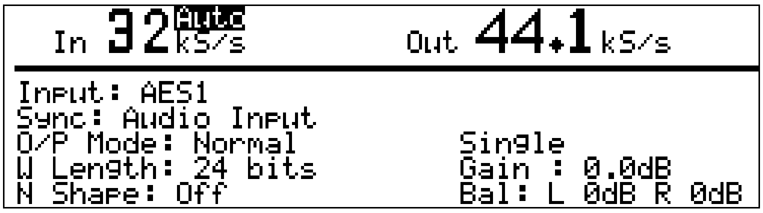
Only select Internal sync if you are using the unit as a signal generator or a reference clock source.

Multiple Channel Sync

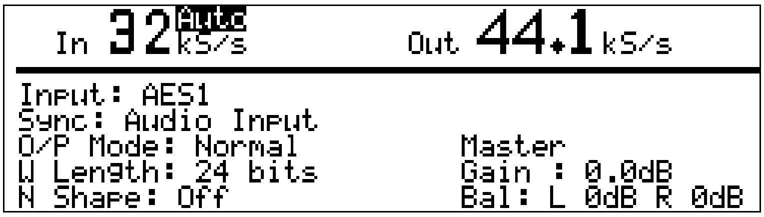
This option is either On or Off.

The unit uses the **AES Ref Loop In** as a sync link. When On, if the unit detects a sync signal into the **AES Ref Loop In**, it will sync to it and set up as a Slave. If there is no sync signal coming into the **AES Ref Loop In**, it will set up as a Master. See section **Multiple Channel Sync'ing**, page **81** for an explanation, or the multi-channel applications from page **29** for wiring. Do not feed a signal with active User bits into **AES Ref Loop In** with Multiple Channel Sync turned on!

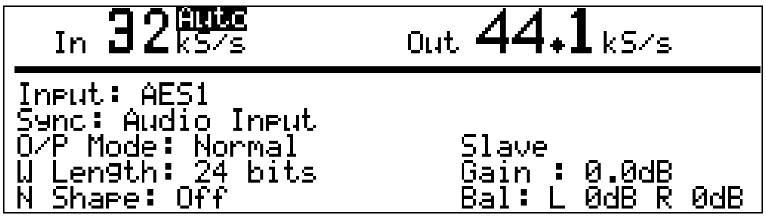
When the option is On, the **Status** display informs you about the unit's configuration. If an active signal is connected to the **AES Ref Loop In**, but it does not carry User bits, the unit thinks it is a **Single** unit, and the **Status** display is as follows:



If no signal is connected into the **AES Ref Loop In** (there is no sync link going in), the unit thinks it is a **Master** and the **Status** display is as follows:



If the sync link is connected and active, the unit thinks it is a **Slave**, and the **Status** display shows:



For multi channel syncing on DSD to DSD, the sync link does not synchronise all the output Wordclocks. Each unit will have a different phase. Just use the **Wordclock** from one unit and ignore the rest – the group delays for the signals will not be affected.

Input Sample Rate

For PCM input modes, the input sample rate can be sensed and set automatically, using the Auto option, or forced. The latter is useful where use of 1 or 2 wire mode for higher sample rates might be ambiguous – otherwise Auto is best. It occurs at the bottom of the Input Sample Rate list:

```
Sample Rate Conversion
    Input Sample Rate
         24kS/s                                  ↑
         22.05kS/s
         16kS/s
         12kS/s
         11.025kS/s
        ■Auto □                                  ↓
```

Select the required entry from the list and press **Set**. If you have selected Auto, then the list changes when re-entered:

```
Sample Rate Conversion
    Input Sample Rate
        ■Auto ⊠
```

and the **Status** display will indicate that Auto is on:

```
  In 96 Auto kS/s            Out 44.1 kS/s
Input: AES1
Sync: Audio Input
O/P Mode: Normal
W Length: 24 bits                Gain : 0.0dB
N Shape: Off                     Bal: L 0dB R 0dB
```

Output Sample Rate

Output Sample Rate is a parameter for PCM output modes only. Select the required Output Sample Rate from the list and press **Set**. The ***dCS 974*** accepts 112 input / output sample rate combinations (including DSD) in one pass. The remaining 70 combinations can be accommodated in 2 passes. These are shown in **Table 2** overleaf.

Not all the apparently valid output sample rates may be available to you from the menu. When Auto Input Sample Rate Selection is turned on, the menu system dynamically alters the Output Sample Rate menu to reflect the valid **1 pass** output rates for the current input rate. For example, if Auto is turned on and the input is running at 96 kS/s, then the Output Sample Rate menu would look like this:

```
Sample Rate Conversion
    Select Valid Output Rate
         192kS/s
         96kS/s
         88.2kS/s
        ■48kS/s
         44.1kS/s
         32kS/s                                  ↓
```

INPUT SAMPLE RATE (kS/s)	OUTPUT SAMPLE RATE (kS/s)												
	11.025	12	16	22.05	24	32	44.1	48	88.2	96	176.4	192	DSD
11.025	1 pass	1 pass	2 pass	1 pass	1 pass	2 pass	1 pass	2 pass	1 pass	2 pass	2 pass	2 pass	2 pass
12	1 pass	1 pass	2 pass	1 pass	1 pass	2 pass	2 pass	1 pass	2 pass	1 pass	2 pass	2 pass	2 pass
16	2 pass	2 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	2 pass	1 pass	2 pass	2 pass	2 pass
22.05	1 pass	1 pass	1 pass	1 pass	1 pass	2 pass	1 pass	1 pass	1 pass	2 pass	2 pass	2 pass	2 pass
24	1 pass	1 pass	1 pass	1 pass	1 pass	2 pass	1 pass	1 pass	2 pass	1 pass	2 pass	2 pass	2 pass
32	2 pass	2 pass	1 pass	2 pass	2 pass	1 pass	1 pass	1 pass	1 pass	1 pass	2 pass	2 pass	2 pass
44.1	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass
48	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass
50	2 pass	2 pass	2 pass	2 pass	2 pass	2 pass	1 pass	1 pass	2 pass	2 pass	2 pass	2 pass	1 pass
88.2	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass
96	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	2 pass	1 pass	1 pass
176.4	2 pass	2 pass	2 pass	2 pass	2 pass	2 pass	1 pass	2 pass	1 pass	2 pass	1 pass	2 pass	1 pass
192	2 pass	2 pass	2 pass	2 pass	2 pass	2 pass	1 pass	1 pass	1 pass	1 pass	2 pass	1 pass	1 pass
DSD	2 pass	2 pass	2 pass	2 pass	2 pass	2 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass	1 pass

Table 2 – One Pass and Two Pass Conversions

If the input sample rate were 32 kS/s, then the display would offer a different selection:

```
Sample Rate Conversion
    Select Valid Output Rate
         96kS/s
         88.2kS/s
         48kS/s
        ■44.1kS/s
         32kS/s
         16kS/s
```

The input sample rate can be changed to one that is incompatible with the currently selected Output Sample Rate, because it needs a 2 pass operation. As an example, we might have 11.025 kS/s coming in (not on Auto), and try to select 96 kS/s. The *dCS 974* will detect that this ought to be a 2 pass operation and display the following information box:

```
        Auto
    I
Inp   Invalid Fs In for Fs Out
Syn   Press a key to alter Fs Out
W L
N S
Filter Cut: 1             Bal: L 0dB R 0dB
```

While displaying this information box, the output of the *dCS 974* will be muted. To continue, either:

do this: Change the Input Sample Rate to one suitable for conversion (see the table above) or

do this: Press any key, and the *dCS 974* will display a list of valid Output Sample Rates, as follows (for our 11.025 kS/s example):

```
Sample Rate Conversion
    Select Valid Output Rate
          88.2kS/s
         ■44.1kS/s
          24kS/s
          22.05kS/s
          12kS/s
          11.025kS/s
```

For a **2 Pass** conversion, convert the input to an intermediate sample rate and either feed it direct into another ***dCS 974*** or record the result, then convert from the intermediate sample rate to the required sample rate. You can always choose one of 44.1 kS/s or 48 kS/s as the intermediate sample rate, so you can record the intermediate pass on most recording devices.

However, it is best to use the highest possible sample rate for the intermediate pass, if your recording device can take it. For example, to convert from 192 kS/s to 12 kS/s, first convert from 192 kS/s to 96 kS/s, and then convert from 96 kS/s to 12 kS/s. Use of two ***dCS 974***'s avoids intermediate storage.

Output Mode

For DSD output, select DSD SDIF-2 or DSD SDIF-3, and the unit will make the appropriate changes. The unit supports DSD out with either DSD or PCM in. DSD output includes a DSD DC blocking filter. All signals in DSD or PCM will have their DC filtered out. When set to either DSD output mode, a DSD Quad signal appears on four output cables – **AES 1, AES 2, AES 3** and **AES 4**. The 3 SPDIF outputs all carry identical AES streams with no data – they may be used for synchronisation.

For PCM output, select Dual AES to output a 192kS/s, 176.4kS/s, 96 kS/s or 88.2 kS/s AES3 signal on two output cables – **AES 1** and **AES 2** (or **AES 3** and **AES 4**). In this mode, the **AES Clk Output** and 3 SPDIF outputs all carry identical AES streams with no data – they may be used for synchronisation. The **SDIF Clk Output** sends Wordclock at the same rate as on **AES 1**, do not use the SDIF data outputs while in Dual AES mode.

For PCM output, select Quad AES to output a 192kS/s or 176.4kS/s AES3 signal on four output cables – **AES 1, AES 2, AES 3** and **AES 4**. In this mode, the **AES Clk Output** and 3 SPDIF outputs all carry identical AES streams with no data – they may be used for synchronisation. The **SDIF Clk Output** sends Wordclock at the same rate as on **AES 1**, do not use the SDIF data outputs while in Quad AES mode.

Otherwise, set to Normal for single wire outputs on **AES 1, AES 2, AES 3** & **AES 4**, the three SPDIF outputs and the SDIF-2 output.

For PCM out, the unit can have PCM or DSD in.

Filter

The ***dCS 974*** offers a choice of filter for use on some of the more popular in/out combinations. The filters offer differing responses. For PCM outputs, Filter 1 in each case offers the sharpest cut-off and no or least aliasing, but longest energy smear. Filter 4 gives the gentlest roll-off (usually with significant aliasing) but the shortest transient response with least smear. For DSD output, the differences are not so great – the responses are given in the section **dCS 974 Performance Curves**, page **94**. The conversions that offer multiple filters are as follows:

In	Out	Number of Filters Available
DSD	DSD	8
PCM	DSD	8
192	96	4
176.4	88.2	4
176.4	44.1	4
96	48	4
96	44.1	4
88.2	44.1	4
48	44.1	4
44.1	96	4

Table 3 – Conversions with Multiple Filter Options

We encourage you to experiment with the filters, to find the one that sounds best for your particular application. Do not assume that one filter is best for all applications![7]

The menu is dynamic - that is, it updates to show valid selections. For example, Fs In = 96kHz, Fs Out = 44.1kHz has 4 options and shows:

```
Sample Rate Conversion
    Filter
        ■Filter cut 1
         Filter cut 2
         Filter cut 3
         Filter cut 4
```

While Fs In = 96kHz, Fs Out = 88.2kHz has only one option and shows:

```
Sample Rate Conversion
    Filter
        ■Filter cut 1
```

The ***dCS 974*** remembers the last filter selection for every conversion, so if you choose Filter 4 for 96⇨44.1 and Filter 2 for 96⇨48, these separate settings will be stored and loaded when you switch between them.

Output Word Length

Select the required Word Length from the list and press **Set**.

IMPORTANT!

*The **dCS 974** generates long word length (24 bit) data and truncating this adds extra noise. It can also add highly undesirable behaviour at low signal levels. We recommend that you use a high order noise shaping function, but see the section* **Word Length Reduction**, *page* **104** *and try for yourself. This is a seriously major topic, and you should experiment.*

[7] The reports we receive from users suggest that for PCM work, Filter 2 is well suited to some classical music, and that Filters 3 and sometimes 4 suit rock. However, views vary quite a lot.

Noise Shaping

Noise Shaping is a technique in which the noise energy added by truncating a longer word is pushed into the less audible parts of the spectrum - giving a useful improvement in perceived noise level. There is a trade-off between noise floor improvements in the mid-band, where the ear is most sensitive, and increased noise at the top of the audio band, which the ear does not hear but your system might. Noise Shaping is achieved by processing the truncated bits so it is not available if the output word length is the same as the input word length. Dither is different (and additional) to Noise Shaping, and is necessary under some circumstances. ***dCS*** consider Dither to be unnecessary in many situations - Noise Shaping alone is sufficient, and lower noise. See the section **Word Length Reduction**, page **104** for more information.

Entering the menu shows:

```
Sample Rate Conversion
   Noise Shaping
      ■Off
       1st Order
       2nd Order
       3rd Order
       4th Order
       5th Order                        ↓
```

Select the required Noise Shaping characteristic from the list and press **Set**. The options are up to 10th order, for sample rates up to 96kS/s. Noise Shaping is not applicable to DSD. For 32 kS/s, 44.1 kS/s, 48 kS/s, 88.2 kS/s and 96 kS/s the curves are individually optimised[8].

Dither

Entering the Dither submenu shows:

```
Sample Rate Conversion
   Dither
      ■Off
       Triangular
       Top Hat
       N. Shaped Triangular
```

Select the required Dither characteristic from the list and press **Set**.

N. Shaped Triangular means Noise Shaped Triangular.

For more information on dither, and different dither types, see the section **Dither** on page **85** and **Figure 31** to **Figure 34**.

[8] If you need curves optimised at other sample rates, contact us.

AES Message Edit

This menu sets the message bits on the AES outputs.

Professional On/Off	Select then press **Set** to toggle between On and Off.
Non-Audio On/Off	Select then press **Set** to toggle between On and Off.
Mode	Select the audio data format then press **Set**. Options are: Not indicated Two channel Primary/Secondary Stereophonic
Source	Select each of the four bytes in turn, press **Set**, turn the rotary control to change the character as necessary and press **Set** again to accept. Press ← to return to the previous menu and the four characters will be displayed beside "Source".
Destination	Set up similarly to Source.

SPDIF Message Edit

This menu sets the message bits on the SPDIF outputs.

Professional On/Off	Select then press **Set** to toggle between On and Off.
Non-Audio On/Off	Select then press **Set** to toggle between On and Off.
Copy Permit On/Off	Select then press **Set** to toggle between On and Off.
Format	Select the audio data format then press **Set**. Options are: 2 Channel General Format Compact Disk 2 Channel Encode/Decode DAT

Gain/Balance and Maximise

This menu allows gain (output level) and balance to be set, or an automatic gain setting to get the highest possible level without overload – for CD preparation, for example. The balance control works in a different way to conventional (analogue) balance controls, and although is easy to operate, you should be aware of what it does.

To set the gain, select Gain then press **Set**. Turn the rotary control to adjust the Gain anywhere between –100.0 dB and +12.0 dB in 0.1 dB steps[9]. Press **Set** again to exit.

[9] The *dCS 974* has a global gain of -0.01 dB absolute, so when the unit says "0 dB" the overall gain will be -0.01 dB. This is because noise shaping, dither, and a few other mechanisms can add (small) amounts to an input signal, but enough to cause digital clipping. The -0.01 dB global gain shift avoids the problem.

IMPORTANT! *Use positive gain with care as the unit does not monitor overloads. If in doubt, use* Maximise *instead.*

With Balance set central (i.e. L 0dB, R 0dB), no gain is applied to either channel. Panning to the right attenuates the left but does **not** amplify the right channel and vice versa. This arrangement helps avoid overloads.

To set the balance, select Balance then press **Set**. Turn the rotary control to adjust the balance anywhere between 0.0 and -6.0dB on either channel in 0.2dB steps. Adjustment beyond -6.0dB mutes that channel. Press **Set** again to exit.

Maximise is a feature that makes best use of the available dynamic range. The ***dCS 974*** keeps a running record of the maximum amplitude of the signal. Maximise can then apply the exact gain to ensure the highest signal peak noted is boosted to full scale.

The maximum gain that may be applied is +12dB. If the correction exceeds this, +12.0dB of gain will be applied. Please note that Maximise cannot correct overloads. Maximise can be run repeatedly if you have a lot of gain to max up.

do this: At the start of the track, select Maximise then Reset max hold and press **Set**.

do this: At the end of the track, select Maximise then Maximise! and press **Set**. The gain correction will be applied.

do this: Replay the track and the gain will be Maximised.

Select Maximise then Display max hold and press **Set** to see the current peak signal record.

Swap Channels

```
Sample Rate Conversion
    Swap Channels
        ■Normal
         Swapped
```

This allows the left and right channels to be swapped. For normal operation, select Normal and press **Set**. To swap channels, select Swapped and press **Set** - the **L/R Flip** LED will light up.

Phase

The options for Phase are:

Normal	Select then press **Set**. Outputs are in phase with inputs. Both **Phase** indicators are off.
Left Inverted	Select then press **Set**. Left channel is out of phase with input, Right is in phase. Left **Phase** indicator lights.
Right Inverted	Select then press **Set**. Right channel is out of phase with input, Left is in phase. Right **Phase** indicator lights.
Both Inverted	Select then press **Set**. Both channel outputs are out of phase with inputs. Both **Phase** indicators light.

Detect Silence

Detect Silence is a PCM only option that gives digital silence out if the unit detects digital silence in. This is useful where dither, and/or noise shaped truncation are being used – without it, even if the signal in goes to digital silence, the output will continue to output low level noise. It allows some automation further along the CD preparation chain. The options are:

On	Select then press **Set**. Output goes silent after about 25 ms of silence in. Output recovers immediately when the input goes live, although it takes about 25 ms for the dither to turn back on if this is used.
Off	Select then press **Set**

IMPORTANT! *Digital Silence detection does not currently operate in DSD modes.*

De/Pre-Emphasis

For Output Sample Rates of 32kS/s to 96kS/s, Emphasis may be applied or removed using this menu. The menu displays:

```
Sample Rate Conversion
   De/Pre-Emphasis
      ■Off
       De-Emphasis:50/15uSec
       De-Emphasis:CCITT J.17
       Pre-Emphasis:50/15uSec
       Pre-Emphasis:CCITT J.17
```

Select the appropriate function, and press **Set**.

IMPORTANT! *Note that the unit drops the signal level by 12dB when applying pre-emphasis, to prevent overloads occurring.*

When removing pre-emphasis (de-emphasising) there is no signal drop. When units are run back to back, therefore – in any order – applying and removing pre-emphasis, there will always be a 12 dB signal drop.

Display Customise

This menu allows the user to determine which five parameters are displayed on the status screen in addition to input and output sample rates. The active parameters in the list are indicated by the **x** sign in the box next to it. For example:

```
Sample Rate Conversion
   Display Customise
      ■☒  Input
       ☒  Sync
       ☒  O/P Mode
       ☒  W Length
       ☒  N Shape
       ☐  Dither                                ↓
```

indicates that the five displayed parameters are Input, Sync, O/P Mode, W Length and N Shape. Dither and the options further down the list are not displayed.

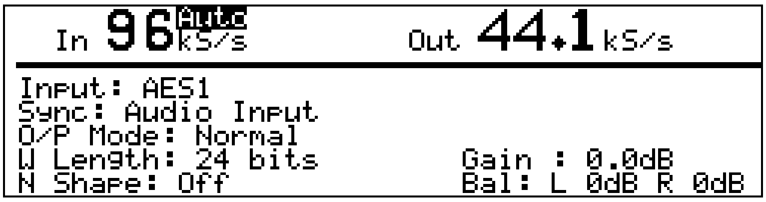
To display a parameter that is not marked, first check that no more than four parameters in the list are marked. If so, select the extra parameter and press **Set** - an **x** sign will appear in the box next to it.

If five parameters are already marked, select an unwanted one and press **Set**; the **x** sign will disappear. Then select the wanted parameter and press **Set**.

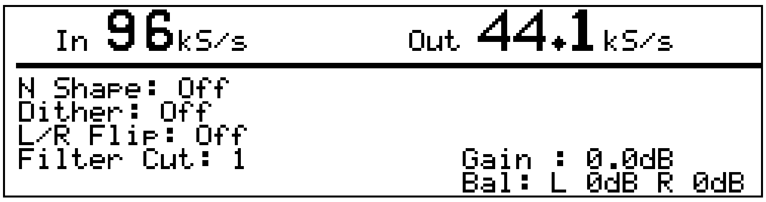
The parameters available are:

Input	Audio input selection
Sync	Synchronisation source
O/P Mode	Output mode
W Length	Word Length
N Shape	Noise Shaping characteristic applied
Dither	Displays Dither type used
L/R Flip	
Filter	Displays Filter Cut used

With the top five parameters selected, the **Status** screen might look like this:



With the bottom four parameters selected, the **Status** screen might look like this:



Error Monitor Submenu

The ***dCS 974*** keeps count of CRC, Parity and Invalid errors and displays them on the Error Monitoring screen. To view this, go to the top level menu, select Error Monitor and press **Status**:

```
              Error Monitoring
CRC Errors:        L 0 R 0
Parity Errors:     L 0 R 0
Invalid Flags:     L 0 R 0
```

This shows the error count on Left and Right channels separately since the last reset.

Error Hold and Reset

Select the Error Monitor menu and press → to display the next level menu:

```
Error Monitor
   ►CRC Errors Left
    CRC Errors Right
    Parity Errors Left
    Parity Errors Right
    Invalid Errors Left
    Invalid Errors Right
    All
```

Press → again to display the options for CRC Error Left:

```
Error Monitor
   CRC Errors Left
      ■Hold
       Reset
```

Select either Hold or Reset and press **Set**. The screen display changes to the Error Monitoring status. Hold stops the count for that error type and Reset sets the count to zero. Each error count may be stopped or reset in this way. The All menu applies the changes to All Errors.

Test Submenu

Generator Overview

The ***dCS 974*** includes a high quality digital signal test generator, with both sine or square wave outputs, and displays its settings on the **Test Mode** screen. When invoked, the generator adjusts its output format to that currently operating, and substitutes the generated output for any input signal. The generator operates at the output sample rate. In PCM modes, it is a PCM generator (in the appropriate mode), in DSD mode it is a DSD generator[10].

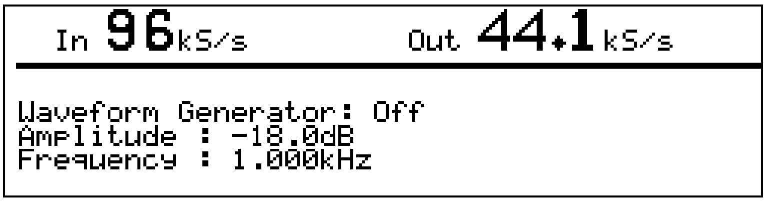
If the unit is sync'd to an input when it goes into Generator mode, then the generator stays sync'd to that input (so the output will be synchronous and locked).

IMPORTANT! *If the unit is not sync'd to anything, set the* Sync Source *to* Internal*. Otherwise, the equipment you are trying to drive may be unable to lock to the Generator outputs because the sample rate has drifted out of lock range.*

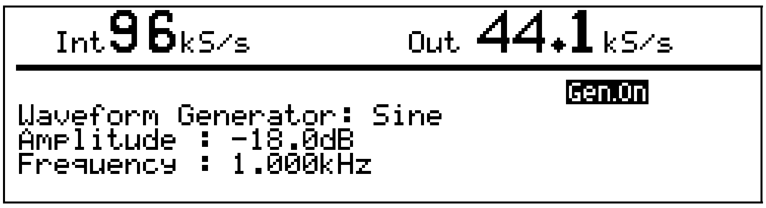
The Sample Rate Conversion menu is still active in Generator mode, so it can enter Generator mode with no external syncing, the Generator can be set up, a sync input can be selected from the Sample Rate Conversion menu and the unit will sync to that. Noise shaping and other features available in the Sample Rate Conversion menu can still be used, as long as these operate at the output rate. Filters, for example, do not affect the generator output, because they operate on the input signal. Dither operates at the output rate, so it can be used.

Controlling the Generator

To view the **Test Mode** screen, go to the top level menu, select Test and press **Status** to get the display below (assumes 96kS/s sync source):

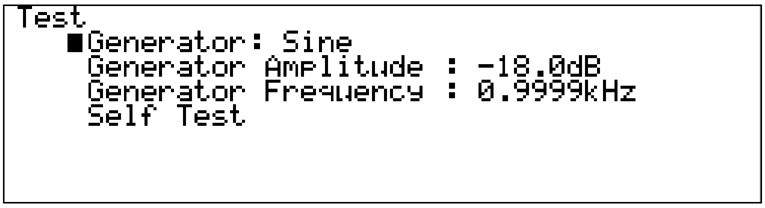


If the unit is set to Internal sync, the screen will show **Int** beside the input sample rate as follows:



10 in DSD mode it is a PCM generator, operating at a sample rate of 705.6 kS/s, and then converted to DSD. This means that the maximum frequency that it can produce is 352.8 kHz.

To set up the generator, press → twice to move to the menu screen then display the **Test** menu:



The Generator options are Off, Sine and Square. Press **Set** slowly to scroll through these. There is a short delay while the generator code is loaded.

IMPORTANT!

*The initial Generator Amplitude is -18dB0. If the **dCS 974** is ultimately driving speakers or headphones, ensure the system gain is at a reasonable level before turning the generator on.*

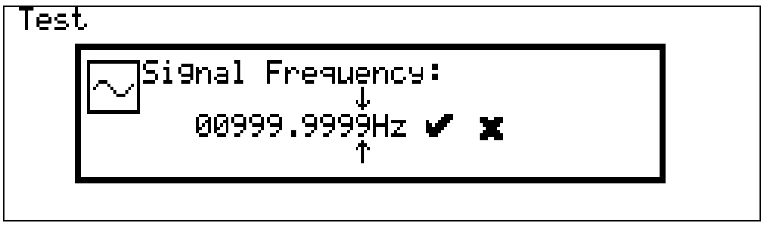
Generator Amplitude Adjustment

Select Generator Amplitude and press **Set** to adjust it. Turn the rotary control counter-clockwise to reduce the amplitude in 0.5dB steps or clockwise to increase it. Attempting to increase the amplitude above 0dB causes the setting to wrap around to -120dB. When the required amplitude is displayed, press **Set** again to update the setting.

Generator Frequency Adjustment

Select Generator Frequency and press **Set** to adjust it. The following screen appears:

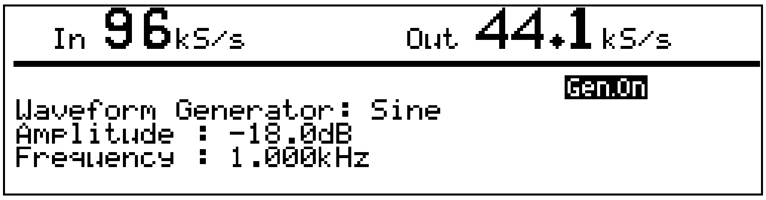
The frequency resolution of the generator is about 1 part in 2^{31} of the sample rate – about 10µHz at 48 kS/s. Rotating the knob moves the arrows left and right. When the arrows point to the digit that you want to edit, press the **Set** button. The knob will now increment or decrement the selected digit. If you increment the digit beyond 9, then the next digit gets incremented, and the current digit is set to 0. Similarly if the digit is at 0, decrementing it will change the previous digit(s). For example, the above display shows 1kHz. If we decrement the .0001 digit, the result is as follows:



When you have set the frequency you want, move the arrows to the ✔ symbol and press **Set**. To discard the changes and return to the tone generator menu, move the arrows to the ✖ symbol and press **Set**.

If you select a frequency that is more than $\frac{outputFs}{2}$, the frequency generator will automatically be set to $\frac{outputFs}{2}$.

Select Generator: Off and press **Set** - it will toggle to Generator: Sine. Wait several seconds for the code to load up, then press again and it will change to Generator: Square. Press again and it will return to Generator: Off. Press **Status** to display the **Test Mode** screen:



The Gen.On label appears on all status screens while the generator is on.

IMPORTANT!

With the Generator On, the signal path from the selected input is disabled and replaced by the Test Generator signal.

The Generator is turned off and reset to -18.0dB, 1kHz at power down.

Self Test

The last option on the test menu is Self Test - this exercises the LEDs and the LCD display. Select this and press **Set**. If all is well, the LEDs will flash in sequence and the LCD will display a pattern of flashing columns. Press any button to exit from this.

Info Submenu

To display information about the unit, return to the top level menu, select Info and press **Set**. The **Info** screen will be displayed:

```
                 Info
Front Panel EEPROM v1.00
Control Board EEPROM v1.00
Control Board S/N:0002-540-061-0024
dCS email : dcsddc@dcsltd.co.uk
PCM Conversions Supported: 98
Conversions with > 1 filter: 9
```

The first three lines give important details about the hardware and software installed in your ***dCS 974***. This is helpful if you experience difficulties with your unit. Line four is ***dCS*** E-mail address for use if you need any further assistance. The last two lines give you a hint of the options available on your machine.

Bit Activity Monitor Submenu

Bit Activity Monitoring Overview

This monitors activity on the selected **input** and indicates the wordlength of the data provided by the source device. Because the monitor works on the input, it can be used for DSD outputs if the input is PCM. To monitor the PCM output, use Level Meters instead.

Setting the Monitor

To access the monitor, return to the top level menu, select Bit Activity Monitor and press **Set** or **Status**. With a substantial 20 bit input signal, the display will be similar to this:

```
        Bit Activity Monitor
MSB                    LSB
 |                      |
L********************----
R********************----
 |       |       |      |
23      15       7      0
```

The display is updated approximately three times per second.

The Display Timeout function is disabled while the Bit Activity Monitor is being used.

Level Meters Submenu

Level Meter Overview

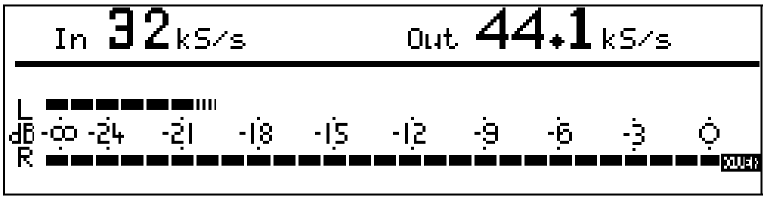
The Level Meters monitor the Left and Right **output** signal levels. They work on all PCM signals, and can be used to monitor DSDby setting a DSD to PCM conversion.

The ***dCS 974*** can be used purely as a programmable meter. Because the meters work on the output data, the filters can be used to change the bandwidth of the meters. For example, they can be used to monitor 96kS/s data with a 20kHz bandwidth, by setting the output rate to 44.1kS/s, or the full bandwidth can be examined by setting the output rate to 96kS/s.

The Display Timeout function is disabled while the Level Meters are being used.

Turning the Level Meters On

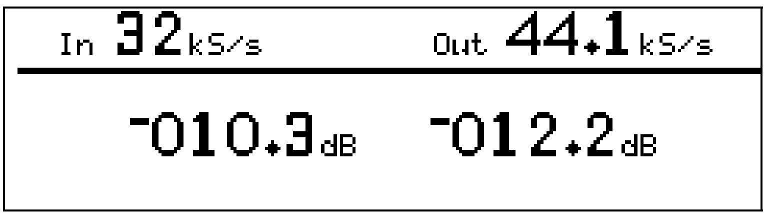
To access the Meters, return to the top level menu, select Level Meters and press **Set** or **Status**. With an input signal applied, the display will look similar to this:



The bar consists of 27 segments, each representing 1dB. The dashed segments at the right end of each bar are the **Peak Hold** indicators.

The indicator on the lower of the two bars shows that overloads[11] (full scale signals) have occurred in the last monitoring period, on that channel (it says "**Over**" if you peer closely). The decay of the overload is controlled by the Peak Hold function in the Level Meters menu (below).

Numerical metering to a resolution of 0.1dB can be set up instead, for example for accurate alignment in comparisons. See Meter Type below to do this – the display will change to:



11 because the ***dCS 974*** reduces the signal level by 0.01dB, this can only occur if gain is applied, or if the unit is in Format Conversion mode and the input signal goes to full scale.

Meter Type (Bar or Numerical)

To change between Numerical or Bar level metering, or to change the meter characteristics, return to the Level Meters menu and press → to display the options:

```
Level Meters
    Decay
    Peak hold
   ■Meter Type: Bars
```

Decay Time

Select Decay and press **Set** to display the Decay time options:

```
Level Meters
   Decay
      ■Normal
       Faster
       Fastest
       Instant
```

These options give the following characteristics:

Normal	the bar length decays away at 32 ms/segment
Faster	the bar length decays away at 16 ms/segment
Fastest	the bar length decays away at 11 ms/segment.
Instant	the bar length decays away completely in 32ms.

Select the required Decay Time and press **Set**.

Peak Hold

Select Peak Hold and press **Set** to display the Peak Hold options:

```
Level Meters
   Peak hold
      ■None
       320mS
       640mS
       960mS
       1280mS
       1600mS                ↓
```

Scroll down to see the bottom of the list of options:

```
Level Meters
   Peak hold
       640mS                 ↑
       960mS
       1280mS
       1600mS
       Infinite
      ■Reset
```

The unit notes the highest signal peak on each channel and turns on the appropriate meter segment on each bar. The **Peak Hold** segment remains for the set time and is then replaced by the next signal peak.

The menu options function as follows:

None	Disables the **Peak Hold** feature.
320ms thru 1600ms	Sets the **Peak Hold** time.
Infinite	Sets an infinite **Peak Hold** time.
Reset	Resets the **Peak Hold** segments to the current peak.

Select the required option and press **Set**. Infinite allows the highest peaks in a track or long passage to be displayed – you might want this to do a manual maximise:

do this: Select Infinite and press **Set.**
do this: At the start of the track, select Reset and press **Set**.
do this: Press **Status** to display the meter.
do this: At the end of the track the highest peaks will be displayed.
do this: Select Reset again and press **Set** to clear the peak readings.

Using the *dCS 974* to monitor a track

If you just want to use the ***dCS 974*** as a level monitoring device, you can control the bandwidth you monitor by setting the output data rate – 44.1kS/s will give approximately 20kHz bandwidth, whereas 96kS/s will give approximately 40 kHz bandwidth.

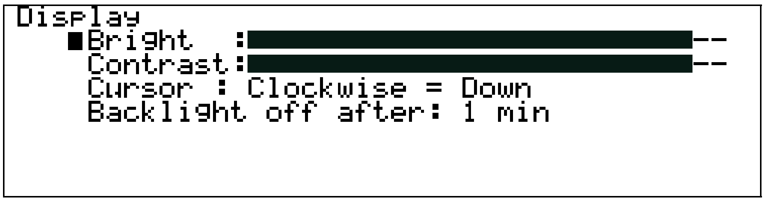
The meters can be used to monitor DSD by setting a DSD to PCM conversion, and using the appropriate output data rate.

Watch Out for this One!

If the unit is tested with sine waves at frequencies which divide almost exactly into the sample rate (e.g. 6.001kHz signal, 48 kS/s sampling frequency), the segments on the right hand side of the meter will flicker. This effect disappears if the signal frequency is changed by about 10Hz and so it is not a problem when monitoring music. It is caused by the signal beating with the sample rate.

Display Submenu

The final option on the top level menu is Display. Select this and press → :



Bright adjusts the brightness of the display backlight and Contrast adjusts the LCD display contrast. These should be set for optimum readability. The length of the solid bars indicate the setting of each parameter - there are 27 levels for each.

To adjust the brightness, select Bright and press **Set**. The cursor will change to the symbol. Set the required brightness with the rotary control then press **Set** again to exit adjustment mode. The Contrast is adjusted in a similar manner.

OOPS!

It is possible to turn the contrast to such a low setting that it is very difficult to read the display. If this happens and the menu level is changed, you may not be able to see the display well enough to navigate to the Display *menu and increase the contrast to readable level. To recover from this you can Recall a stored setup known to have a higher contrast setting (see the next section). If the worst happens and none are available, switch off the unit then hold down the* **Status** *button while switching on. The unit will display* **Using Default Settings** *and will load a setup with high contrast.*

The third option (Cursor) controls the way the cursor responds to turning the encoder. The default setting is the cursor moves down if the encoder is turned clockwise. If you prefer the opposite arrangement, move the cursor to the bottom line and press **Set**. "Clockwise = Down" changes to "Clockwise = Up". Pressing **Set** again change back.

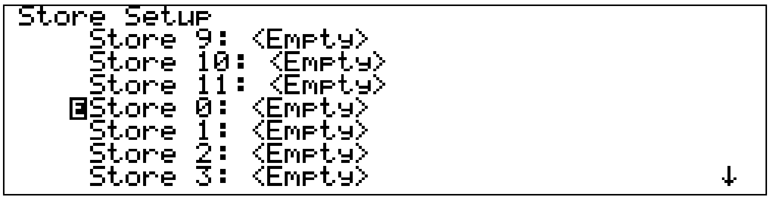
The fourth option determines the time after which the display backlight goes out. The backlight is an electroluminescent device, and so in theory has limited life. It can also buzz a little, and although we select for low buzz, and acoustically damp it, it can be irritating. So – the display goes dark after a preset period, although pressing any button or turning the knob will wake it up again. When the cursor is against the Backlight Off After option, each time you press **Set** it steps through the settings 1 min, 3 min and 5 min.

Setups and Locking the Front Panel

Storing a Setup

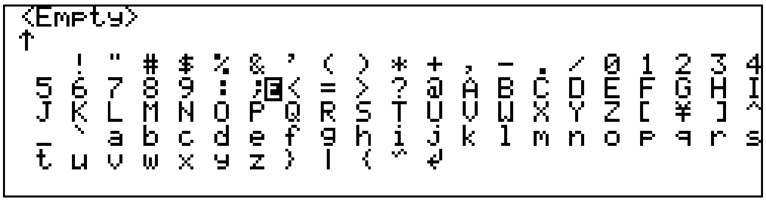
The ***dCS 974*** features 12 non-volatile user programmable setup stores, labelled Store 0 to Store 11. These are accessed by the rotary control and the block of four **MEMORY** buttons on the lower left hand corner of the unit.

First set up the unit as required then press the **Store** button. If nothing has been stored yet, the display will look like this:



You can scroll up and down the list using the rotary encoder and reveal the other five stores. Scroll to the required store (e.g. Store 4) and press the **Edit** button. The display changes to:

To edit the first character in the name, press the **Edit** button again. The display changes - an alphanumeric screen is displayed, with the current store name at the top, and all of the legal characters below it, as follows:



Turn the knob to move the edit block ■ backwards and forwards through the available characters. The arrow ↑ indicates which character you are editing, and the character is updated in real time as you move the edit cursor.

The ↵ symbol represents <End>. When you have finished editing the character, press the **Edit** button to return to the previous screen and select the next character to edit. Names can be up to 15 characters long, plus an <End> character to mark the end of the name. If all 15 characters are used and you attempt to edit the <End> character in the 16th slot, the **Edit** button has no effect. Inserting <End> half way through a name will complete the name there, deleting the right hand portion.

If Store 4 has been named **Demo**, the display will now look similar to this:

```
Store Setup
      Store 1: <Empty>
      Store 2: <Empty>
      Store 3: <Empty>
     ■Store 4: Demo
      Store 5: <Empty>
      Store 6: <Empty>
      Store 7: <Empty>                 ↓
```

Press the **Enter** button and a message window will appear in the display to confirm that the unit is storing the setup.

If you attempt to store a setup in a previously unused location without specifying a name, a message window appears for two seconds, stating **"First choose a name!"**.

You can name your stored setups as anything you like provided the name is no longer than 15 characters. If you have a fixed routine, a few words may be enough to describe the setup. If the unit is used for many different functions by several people, we suggest using the following abbreviations to describe the setup.

Audio Input	use **AES1**, **AES2**, **RCA**, etc, **DA** for Dual AES, **QA** for Quad AES, **DSD** for DSD SDIF, **DSDQ** for DSD Quad.
Input Rate	use **Au** for Auto, **44** for 44.1kS/s, etc.
Sync source	leave blank if synced to Audio Input or use **AR** for AES Reference, **MC** for Multiple Channel Sync, **WC** for Wordclock or **Int** for Internal.
Output Rate	use **44** for 44.1kS/s, or **2496** for 24 bits 96kS/s, or **CD** for 16 bits 44.1kS/s.
Output Mode	leave blank for Normal, **DA** for Dual AES, **QA** for Quad AES, **DSD2** for DSD SDIF-2 or **DSD3** for DSD SDIF3.

Fixed Setups

In addition to the 12 user setups, there are 12 fixed setups which cannot be changed. These are contained in the software and may be added to in future software updates.

Store A	**AES1 Au>24 44** Contains default settings: Auto input on **AES1**, sync to Audio Input, 24 bits 44.1kS/s output in Normal mode. If you get the unit into a tangle, recall this and sort things out.
Store B	**DA Au>CD N9 F2** Auto Dual AES input to CD format, 9th order Noise Shaping, Filter 2, Detect silence On, Gain set to -0.1dB.
Store C	**QA Au>CD N9 F2** Auto Quad AES input to CD format, 9th order Noise Shaping, Filter 2, Detect silence On, Gain set to -0.1dB
Store D	**DSD2>CD N9 F2** DSD SDIF-2 input to CD format, 9th order Noise Shaping, Filter 2, Detect silence On, Gain set to -0.1dB.
Store E	**AES1 Au MC>DSD2** Auto input on AES1, Multi-Channel Sync On, to DSD SDIF-2, Filter 5.
Store F	**AES1 Au MC>24 96** Auto input on AES1, Multi-Channel Sync On, to 24/96 single wire.
Store G	**AES1 Au>192DA** Auto input on AES1 to 24/192 Dual AES.
Store H	**AES1 Au>DSD2 F5** Auto input on AES1 to DSD SDIF-2, Filter 5.
Store I	**QA Au>DSD2 F5** Auto Quad AES input to DSD SDIF-2, Filter 5.
Store J	**DA Au>24 96** Auto Dual AES input to single wire 24/96.
Store K	**AES1 Au>CD N9F2** Auto input on AES1 to CD format, 9th order Noise Shaping, Filter 2, Detect silence On, Gain set to -0.1dB.
Store L	**AES1 Au FC>DA96** Auto (96kS/s) input on AES1 to Dual 24/96, Format Conversion On.

Recalling a Setup

Stored setups may be recalled from memory at any time. Press the **Recall** button once to display the list of user-defined setups:

```
Recall Setup-Press again for fixed stores
     Store 9: <Empty>
     Store 10: <Empty>
     Store 11: <Empty>
    ■Store 0: <Empty>
     Store 1: <Empty>
     Store 2: <Empty>
     Store 3: <Empty>                     ↓
```

Either select one of these or press the **Recall** button again to toggle between the user-defined and fixed setups:

```
Recall Setup-Press again for user stores
     Store J: DA Au>24 96
     Store K: AES1 Au>CD N9 F2
     Store L: AES1 Au FC>DA 96
    ■Store A: AES Au>24 44
     Store B: DA Au>CD N9 F2
     Store C: QA Au>CD N9 F2
     Store D: DSD2>CDN9 F2                ↓
```

Scroll up or down to the required store and press the **Enter** button. A message window will appear in the display to confirm that the unit is reading the setup. When complete, the display shows the **Status** screen for the current menu.

If you make a mistake during a **Store** or **Recall** routine and you have not yet pressed **Enter**, you can abort it by pressing any of the 4 **Operation** buttons.

Locking Out Changes, and Unlocking Again

Once the unit is set up, it may be locked against casual interference by holding down the **Status** button and pressing the **Store** button.

Lockout appears on the right hand side of the **Status**, **Error Monitor**, **Test Mode**, **Info**, **Bit Activity Monitor** or **Level Meter** screen as appropriate and the front panel controls have no effect.

To restore normal operation, hold down the **Status** button and press the **Edit** button. The Lockout label will disappear and normal operation will be resumed.

The Hardware – Controls and Connectors

Rear Panel

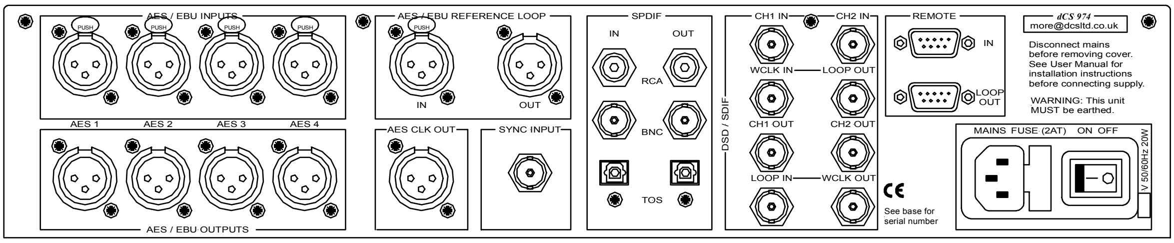


Figure 17– Rear Panel

All input and output connectors are mounted on the rear panel. Individual connectors are identified by the panel legend.

Signal Inputs

AES/EBU Digital Inputs **3 pin XLR Female (4 off – AES 1 – AES 4)**

Used for AES3 format signals. **AES 1** and **AES 2** are used together in **Dual AES** mode. All four inputs are used together for **Quad AES** or **DSD Quad** modes.

AES/EBU Reference Loop **3 pin XLR Female / Male (AES Ref Loop IN & OUT)**

Loop IN is used for AES3 references and also for Sync Link operation in multi-channel synchronising. **Loop OUT** is hard-wired to **Loop IN**, allowing the reference to be daisy-chained through several units.

Sync Input **BNC**

This is intended for future enhancements.

SPDIF Inputs **Various (3 off – RCA IN, BNC IN, TOS IN)**

Used for SPDIF inputs. Pull the plastic dust cover out of the **TOS** input before use.

DSD/SDIF Input **BNC (4 off – CH1 IN, CH2 IN, WCLK IN, LOOP OUT)**

The upper group of four DSD/SDIF connectors form an input interface. They are used as a set for SDIF-2 (PCM or DSD) or SDIF-3 (DSD only).

WCLK IN may be used to sync the unit to an external reference wordclock while taking data from another input. **WCLK IN** is internally wired to **LOOP OUT**, allowing a reference wordclock to be daisy-chained through several units. When the daisy-chain is not used, the **LOOP OUT** connector must be fitted with a 75Ω BNC terminator. **WCLK IN** will also accept a 10 MHz GPS reference.

Signal Outputs

AES/EBU Digital Outputs **3 pin XLR Male (4 off – AES 1, AES 2, AES 3, AES 4)**

Used for AES3 format signals. **AES 1** and **AES 2** (or **AES 3** and **AES 4**) are used together in Dual AES mode. All four outputs are used together for Quad AES or DSD Quad modes. In single AES modes, **AES 2, AES 3** and **AES 4** outputs follow the **AES 1** output.

AES/EBU Clock Output **3 pin XLR Male (1 off – AES CLK OUT)**

AES CLK OUT carries an AES3 clock signal at the same sample rate as the **AES 1** output. It is used for synchronising equipment connected to the outputs of the unit.

SPDIF Outputs **Various (3 off - RCA OUT, BNC OUT, TOS OUT)**

Used for SPDIF outputs. Pull the plastic dust cover out of the **TOS** output before use.

DSD/SDIF Output **BNC (3 off – CH1 OUT, CH2 OUT, WCLK OUT)**

The lower group of three DSD/SDIF connectors form an output interface. Used as a set for SDIF-2 (PCM or DSD) or SDIF-3 (DSD only) outputs.

WCLK OUT carries Wordclock at the same rate as that appearing on the **AES 1** output. It may be used to synchronise equipment connected to any of the *dCS 974* data outputs.

(The lower **LOOP IN** connector is intended for future enhancements.)

Control and Power

Remote **9 pin D type Female (2 off, Remote In and Loop Out)**

For remote control via a PC, and/or downloading software updates. **Remote In** and **Loop Out** sockets allow daisy chaining of several different *dCS* units from one PC COM port.

Mains Supply **3 pin IEC (CEE22)**

Switched, fused and filtered IEC power inlet, for a 50 or 60 Hz AC supply.

Additional Information

As well as connectors, the rear panel displays the following information about the unit, near the mains supply connector:

Mains Voltage The actual voltage setting supplied.
Model Number *dCS 974*
Manufacturers Name and Country of origin (*dCS* Ltd, UK)

The underside of the unit will have a label on that contains a number such as 974-4B1-6B2-2A1-3A2-12345. This is the unit serial number, but it also contains vital configuration information. We will need this number (all of it) to give you support over the phone, or to ship software updates to you.

Front Panel

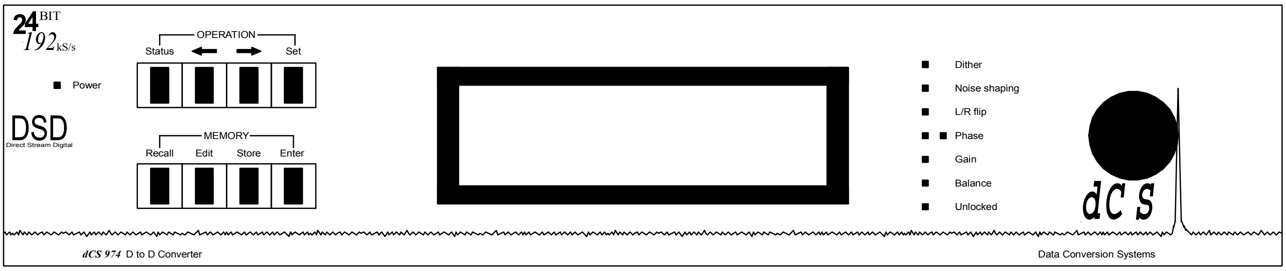


Figure 18 – Front Panel

Because of the many options on the ***dCS 974***, we have used a menu based system to control it. The hardware buttons and knobs below navigate you through it.

Power Indicator

The LED in the top left hand corner lights when power is applied.

OPERATION buttons

The bank of 4 buttons in the top left hand corner are the **Operation** buttons and are used for navigating around the menu tree:

Status	returns the display to the status information screen to allow a rapid check of key settings.
←	selects the previous (higher) menu level.
→	selects the next (lower) menu level.
Set	implements the selected menu item.

MEMORY buttons

The bank of 4 buttons in the lower left hand corner are the **Memory** buttons and are used for storing and retrieving setups:

Recall	allows one of ten stored setups to be selected ready for loading.
Store	allows one of ten locations to be selected as the location for saving the current setup.
Edit	is used to change the name of a saved setup.
Enter	loads or saves a setup.

LCD display

The backlit LCD display in the centre of the panel can display up to 8 lines of information.

LED indicators

A group of LED indicators to the right of the LCD display gives a level of status indication:

Dither	is lit if any Dither is selected.
Noise shaping	is lit if any Noise Shaping is selected.
L/R flip	indicates that the channels are flipped.
Two **Phase** LEDs	indicate if either or both channels are phase inverted.
Gain	is lit if the setting is other than unity gain (i.e. 0dB).
Balance	is lit if the channel balance is shifted from centre.
Unlocked	is lit if the unit is not locked to a valid digital input.

Rotary encoder

Turn the knob to move up down the menu tree, adjust parameter values or edit store names. You may select whether clockwise rotation moves the cursor up or down - see the section on **Display Customise,** page **50.** The unit is shipped with the cursor moving downwards by clockwise rotation of the knob and all instructions in this manual refer to a unit in this configuration. We recommend turning the knob at a steady speed, rather than trying to turn it as fast as possible.

dCS 974 TECHNICAL INFORMATION

Digital Data Formats Supported

dCS 974 provides nine digital data i/o formats:

AES/EBU	(often referred to as AES3) for PCM operation
Dual AES	(part of the AES3 spec) for PCM operation
Quad AES	(part of the AES3 spec) for PCM operation
SDIF-2	for PCM operation
SDIF-2	for DSD operation
SDIF-3	for DSD operation
DSD Quad	for DSD operation
SPDIF	(electrical) for PCM operation
SPDIF	(optical) for PCM operation

For all formats, the incoming Channel Status and User messages are discarded[12]. The unit allows the AES/EBU and SPDIF output Channel Status bits to be edited.

The enhanced AES/EBU interface is fully implemented. Each channel has its own parity and data validity bit, as well as User and Channel Status messages. Cyclic Redundancy Counts (CRC's) are generated from the Channel Status message.

The Dual AES interface allows an 88.2 or 96kS/s 24 bit signal to be coded as two standard 44.1 or 48kS/s 24 bit AES data streams, recorded as four tracks on a recorder with standard capacity, replayed and decoded back into a single data stream. Operation of the Dual AES interface at double speed allows the unit to input or output 2 wire 176.4 or 192 kS/s 24 bit data, and convert to and from this.

The Quad AES interface allows an 176.4 or 192kS/s 24 bit Dual AES data stream to be coded as four standard 44.1 or 48kS/s 24 bit AES data streams, recorded as eight tracks on a recorder with standard capacity. It may be replayed and converted into a Dual AES stream, a single wire format or DSD.

SDIF-2 PCM message bits are internally set to zero, with the exception of the block code, which is implemented.

The SPDIF interface has no CRC's - as per definition. Data formats for both SPDIF electrical and SPDIF optical are identical.

DSD has, at the time of writing, no messaging structure. Contact ***dCS*** for more details. Data formats use either the SDIF-2 system (two data channels and third clock channel), the SDIF-3 format (two data channels with embedded clock) or the DSD Quad format (four AES3 style data streams).

12 At present we do this because there is no standard on what to do with the excess or shortage of bits that is created by a sample rate change. If this causes you a problem, call us – we can probably do something else, if we are clear what that ought to be.

DSD

Filter Options

DSD is a single bit very high sample rate (2.822 MS/s) format, where the single bit words are heavily noise shaped to push noise energy above the audio band. The frequency response is very high (well above 100 kHz) although at these high frequencies, noise is also present.

For SACD purposes, 0 $dB0_{DSD}$ is set at 6 dB below the peak to peak level one might expect a full scale sinewave to occupy – this ensures that artefacts that begin to occur at the limits of the DSD amplitude range do not move down into the audio band. The 0 $dB0_{DSD}$ level is shown graphically in **Figure 39**, page **98**.

The ***dCS 974*** offers a number of different DSD modulators – as Filter options. All the modulators have the same signal frequency response. They differ in the way they shape the out-of-band Q noise, and in how far they suppress the in-band Q noise. Filters 1 to 5 suppress Q noise at least 120 dB below the nominal 0dB DSD signal, which is one of the marketing specs for SACD.

Filter	Comments	SQNR (20 kHz, dB)	Stability	Description
1	High SQNR, high stability	126.14	$1.7*10^{10}$	Two complex zeros
2	High SQNR	127.23	$8.2*10^{8}$	Two complex zeros
3	High SQNR, very high stability	124.66	$1.5*10^{11}$	Two complex zeros
4	Extremely high stability	122.07	$3.0*10^{12}$	Two complex zeros
5	Reduced 100k noise	122.27	$7.1*10^{10}$	Two complex zeros
6	Single complex zero	110.78	$3.7*10^{11}$	Single complex zero
7	Real zeros	101.5	$2.0*10^{10}$	Real zeros only

Table 4 – DSD Filter Summary

Signal to Q Noise and SACD Specs

Although 120dB SNR over the 0-20 kHz band is a good target, it does not match the ears response that well. The F weighted curve is currently accepted as a good model for the ear, and we can use this to weight the noise produced by the various filter choices. Such a weighting is shown in **Figure 38** on page **98**, and DSD gives very good performance using such a weighting (better than 23 bit pcm)

The figure shows that Filter 6 and 7 give more F weighted Q noise suppression in the audio band than Filters 1 to 5. Filter 6 gives around 20 dB more suppression than Filters 1 to 5 under all circumstances and Filter 7 gives 20dB more suppression below 10 kHz.

SACD[13] specifies ultrasonic noise in two bands. These are specs informative specs E2 and E3, and filter performance for a number of specs including these are given below. The measurements given have been made using a Fourier Transform based method.

Filter	Comments	SQNR (20 kHz, dB)	SQNR (F weighted dB)	E2 spec (dB)	E3 spec (dB)
1	High SQNR, high stability	126.14	-136.56	-25.98	-28.96
2	High SQNR	127.23	-138.76	-25.85	-28.95
3	High SQNR, very high stability	124.66	-134.38	-27.32	-30.49
4	Extremely high stability	122.07	-129.91	-25.54	-29.53
5	Reduced 100k noise	122.27	-130.14	-27.13	-31.99
6	Single complex zero	110.78	-151.02	-25.19	-23.55
7	Real zeros	101.5	-132.41	-25.29	-27.04
	SACD Spec			-20.00	-28.00

Table 5 – DSD Filter Performance

DSD Data Formats

DSD is supported on SDIF-2, SDIF-3 and DSD Quad formats.

[13] Super Audio CD System Specifications, Part 2, Audio Specifications, available from Philips System Standards and Licensing, Licensing Support, Building SFF-8, PO Box 80002, 5600 JB Eindhoven, The Netherlands.

PCM Input and/or Output Performance

Filtering

The normal filtering considerations of passband ripple, cutoff frequency and rate, out of band (stop band) suppression, and transient/phase response apply to sample rate converters.

The ***dCS 974*** uses linear phase FIR filters to avoid the limit cycle problems that come with many IIR filters. Linear phase gives filters a symmetrical transient response before and after a transient ("pre-ringing" and "post ringing"). The passband may or may not have a ripple[14], depending on the conversion and/or filter being used. Cutoff frequency is >40% of the lowest sampling rate used in the conversion (input or output).[15] The stop band is typically below –110 dB, but varies with conversion, and can be as low as –130 dB. The frequency responses of two commonly used conversions (96kS/s ⇒ 44.1kS/s and 48kS/s ⇒ 44.1kS/s) are shown in **Figure 41** and **Figure 40** on page **99**.

The transient responses of the filters in the 96 kS/s ⇒ 44.1 kS/s conversion are shown below.

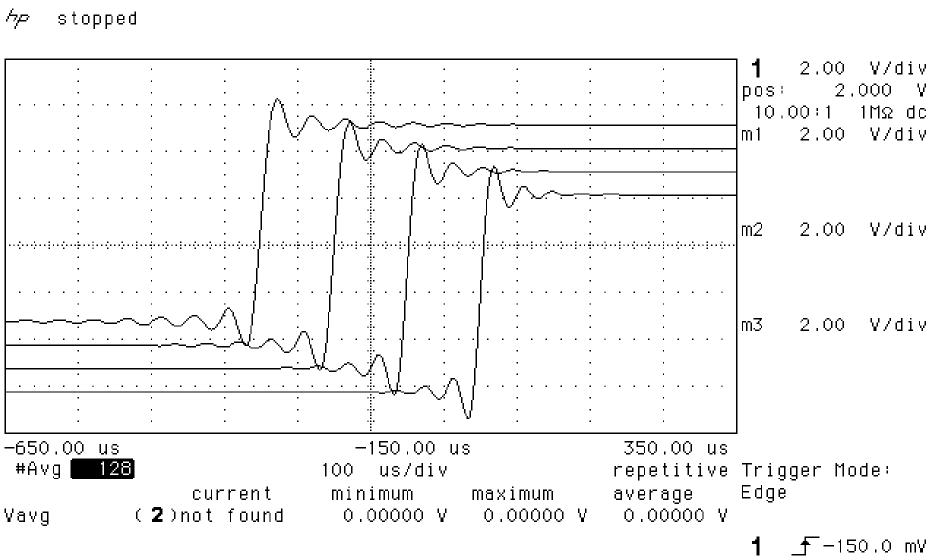


Figure 19 – Transient Performance of 44.1 kS/s Filter Options

Use of higher sample rates gives much tighter transient response, with much less energy smeared into the pre and post ringing. The transient responses of the 96 kS/s output filters are shown below:

14 Filters always have some ripple. For "zero ripple" filters this is in the µdB to pdB region.

15 For conversions where the lowest rate used is 16 kS/s, 22.05 kS/s or 24 kS/s some conversions cut off at 33% of the lowest sample rate.

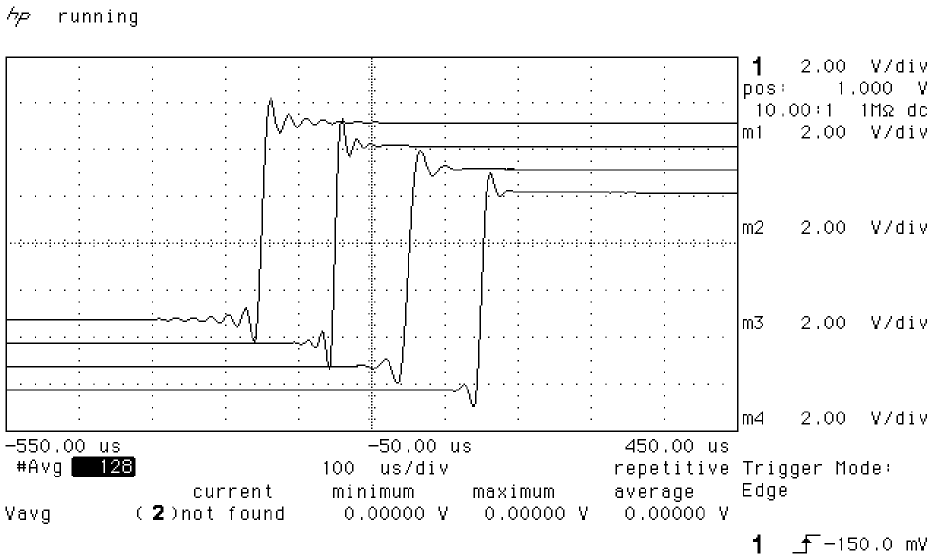


Figure 20 – Transient Performance of 96 kS/s Filter Options

Spurs

Sample rate conversion is a linear process, so spurs are not caused by mechanisms that produce harmonic distortion. Spurs[16] are caused by high order images of the baseband input signal aliasing back into the output[17]. The filtering in the converter has to suppress these.

We measure these spurs for a 1 kHz input signal – typical values are given in dB0 for a -1dB0 sine wave in the table below. The blank cells are two pass conversions, where the performance depends on both conversions.

Input Rate (kS/s)	**Output Sample Rate (kS/s)**											
	11.025	12	16	22.05	24	32	44.1	48	88.2	96	176.4	192
11.025	-130	-122		-143	-117		-136		-140			
12	-139	-136		-131	-135			-138		-137		
16			-132	-119	-132	-119	-112	-130		-137		
22.05	-137	-145	-124	-134	-130		-134	-113	-139			
24	-121	-134	-127	-142	-137		-128	-129		-134		
32			-128			-133	-123	-131	-107	-134		
44.1	-132	-120	-124	-133	-145	-125	-136	-126	-127	-109	-132	-119
48	-118	-132	-136	-119	-134	-125	-137	-136	-127	-130		-132
88.2	-129	-120	-119	-135	-120	-125	-137	-145	-139	-121	-117	-105
96	-97	-126	-138	-116	-135	-139	-117	-137	-119	-139		-117
176.4							-137		-134		-135	
192							-128	-130	-139	-131		-135

Table 6 – Typical Spurious Level vs Conversion

16 Spurious products
17 For example, a 1 kHz tone with a 48 kS/s sample rate in will produce spurs at 2.9 kHz and 4.9 kHz, 6.8 kHz and 8.8 kHz, etc, with a 44.1 kS/s sample rate out.

Noise

Noise arises from processing rounding errors, and from stop band performance. It is typically around –130dB0, but depends on the conversion being used.

Group Delay

The conversion process takes time, and the time is dependent on the conversion being carried out, and the filters used. Typical values in microseconds are given for four conversions in the table below.

Conversion	Group Delay (µs)				Sample Period (µs)	
	Filter 1	Filter 2	Filter 3	Filter 4	Input	Output
44.1k ⇒ 96k	1141	1131	1131	1131	22.676	10.417
88.2k ⇒ 44.1k	1024				11.338	22.676
96k ⇒ 44.1k	1515	1065	1065	1065	10.417	22.676
96k ⇒ 48k	868	958	1028	948	10.417	20.833

Table 7 – Group Delays, PCM to PCM transitions

Clocking

The sample clock quality significantly determines the output performance of a DDC, in as far as items connected may have to reconstitute an analogue signal, and may ultimately derive their clock from the DDC clock.

The highest quality clocks that are available are crystals, so we use these. In Internal sync mode, the ***dCS 974*** uses one of two on-board voltage controlled crystal oscillators (VCXOs) as clock sources – one for 48 kS/s related outputs and one for 44.1 kS/s related outputs.

The internal VCXO is synchronised to the sync source (which need not be the signal input) by a phase locked loop (PLL). The PLL is of a special narrow bandwidth type, that provides a significant degree of "clock cleaning" - but even so, signal quality may degrade if particularly poor slave clocks are used. A consequence of the narrow bandwidth is that it takes quite a long time for the PLL to lock to a new clock frequency – of the order of 2 seconds. The PLL uses DSP assistance to keep this time acceptable.

The input data is extracted using an oversampled UART type of decoder. This can tolerate quite jittery inputs, whose phase is unrelated to the clock source used for locking. The average frequency must be the same, however.

Internal clock

Accuracy when shipped	± 10 ppm
Long Term Stability	± 10 ppm/year at room temperature
Temperature Stability	± 15 ppm over operating temperature range

Synchronising to source

Pull in range	± 300 ppm about nominal frequency
Lock in time	< 2 seconds for most combinations

The PLL is very robust, and will lock to very poor signals if necessary. Data is decoded using a much wider band (faster) PLL, so AES3 type low frequency jitter on the input clock can be handled, and will be cleaned.

If you need to synchronise several items of digital equipment, we recommend using a ***dCS 992*** Master Clock.

Sample Alignment

The ***dCS 974*** aligns samples such that **SDIF WCLK OUT** aligns with AES3 samples out, the rising edge of wordclock aligning with the start of the first illegal code in the X,Z subframe preamble and the falling edge aligning with the start of the Y subframe preamble.

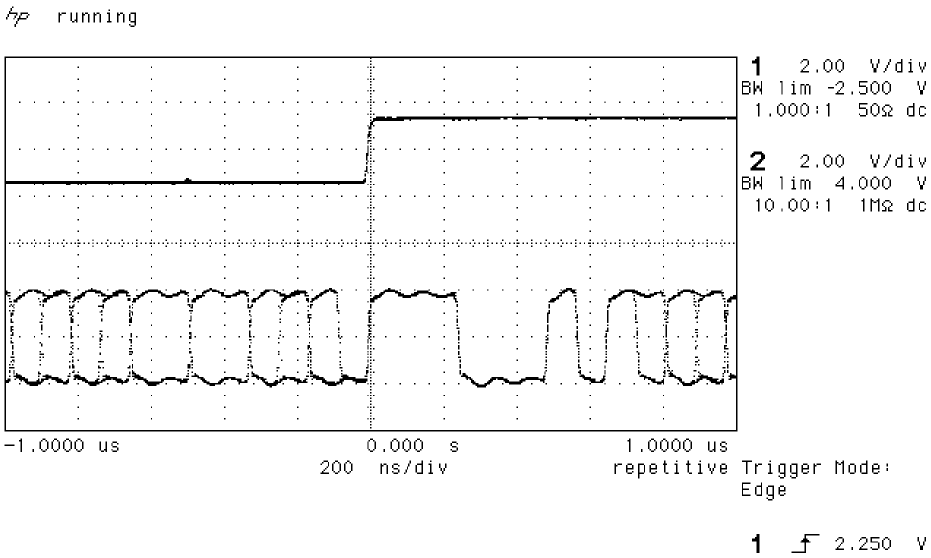


Figure 21 – Wordclock and AES3 outputs, 96 kS/s

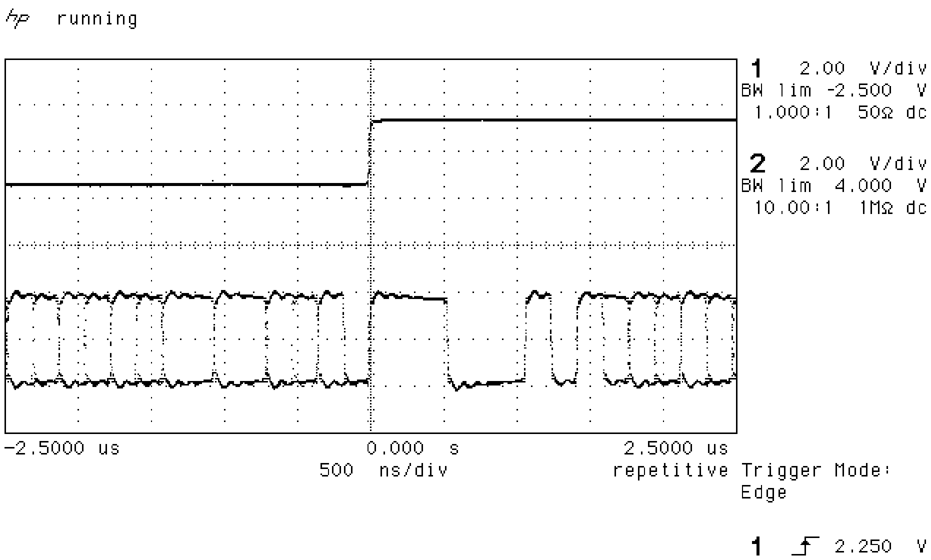


Figure 22 – Wordclock and AES3 outputs, 44.1 kS/s

When **SDIF WCLK IN** is used as a sync source, in and out are related as below. The lower waveform is the output, the upper one is the input.[18] For 44.1 kS/s, out leads in by about 500 ns, and for 96 kS/s it is about 230 ns.

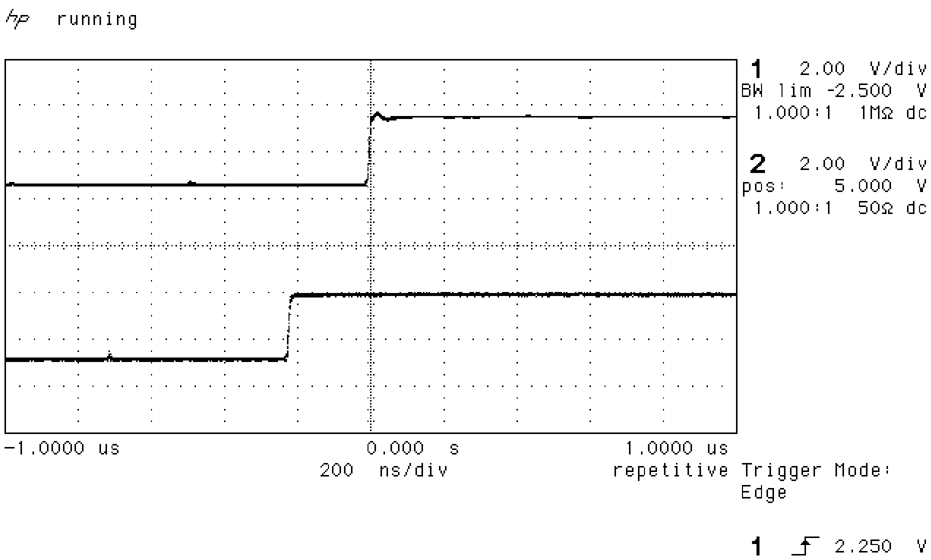


Figure 23 – Wordclock in to Wordclock out, 96 kS/s

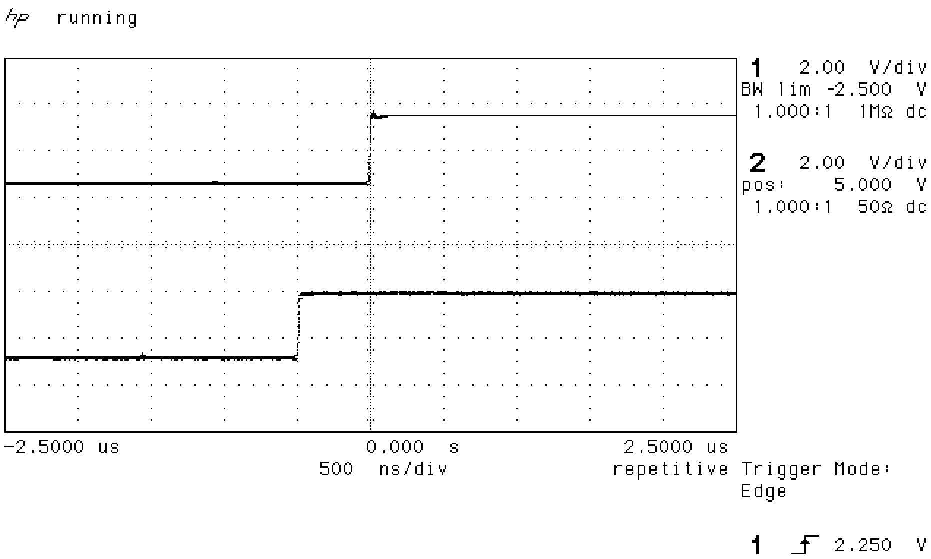


Figure 24 – Wordclock in to Wordclock out, 44.1 kS/s

If tighter alignment of **WCLK IN** to **WCLK OUT** is needed, consider using a ***dCS 992*** Master Clock, where the phase of individual word clock feeds is adjustable, or contact us.

18 If this misalignment causes you a problem, please contact us.

AES3 in and out are related as below, where they are at the same sample rate, and the AES3 input is used as a sync source. The alignment is better than 40ns. Input is at the top of the displays, output is at the bottom. Signals are at the sockets on the ***dCS 974***.

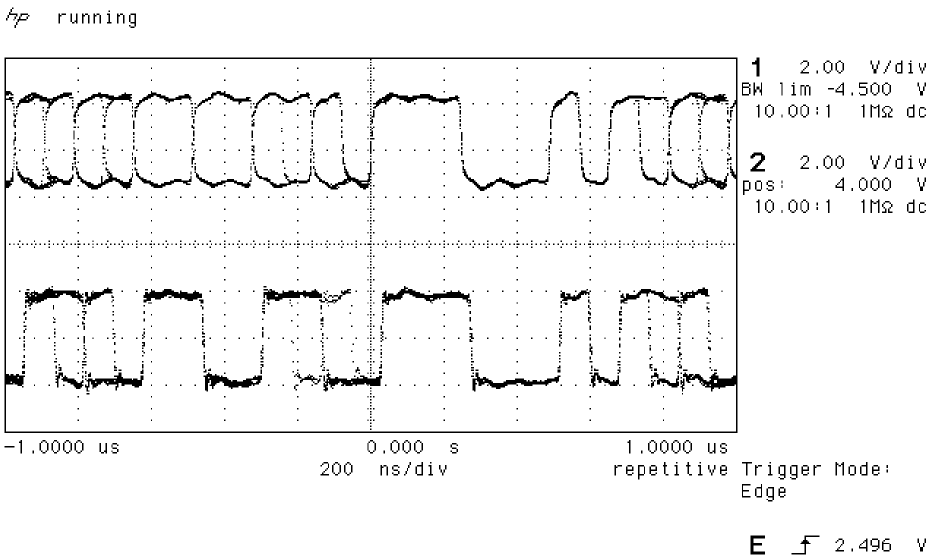


Figure 25 – AES3 in to AES3 out, 96 kS/s

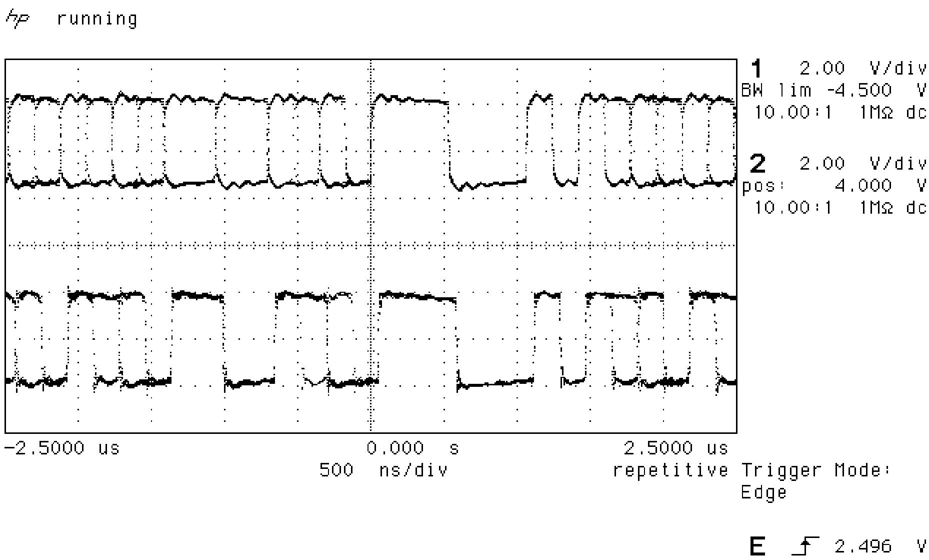


Figure 26 – AES3 in to AES3 out, 44.1 kS/s

DSD data out is aligned to Wordclock transitions, as below – the wordclock edges align with the data transitions, rather than in the middle of the data eye. If you need them the other way round, use an additional 10m of cable in the **WCLK OUT** path.

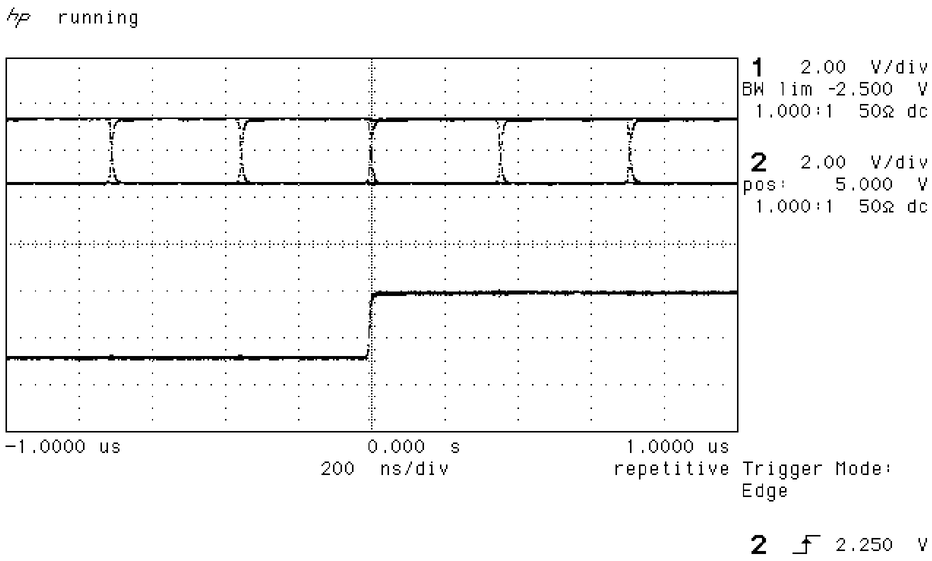


Figure 27 – DSD data and Wordclock out

Multiple Channel Sync'ing

Multiple channel sync'ing requires more things to be in sync than two channel work - the PLL must not be dual modulus, the DSP algorithm must load with a constant group delay, and the frame and block syncs must agree.

The ***dCS 974*** meets all these constraints with the addition of a syncing signal transmitted from a master unit to subsequent slave units. To do this, it passes information in the User Bits of the AES3 message. To slave adequately, units must have this sync information input to the **AES Ref Loop Input**, via a sync link. Each slave will generate a copy, based on its own timing, for transmission to the next unit in the chain. If the Multi-channel Sync option is set to On in the Sample Rate Conversion menu, any units with no user bit synchronisation data into its **AES Ref Loop Input** will become a Master. If user bit synchronisation information is fed in, it will become a slave. Either unit will output synchronisation information via its **AES CLK Output**. A master clock (***dCS 992*** [19]) can be used as the master – if more than about 8 channels are being used, this system is best, as it avoids any tolerance build up. Connections are shown in the multi-channel syncing applications, starting on page **29**.

Using the multi-channel sync mode, the timing of the data at the output of the ***dCS 974*** is related to the data coming in. If three ***dCS 974*** units are used in a six channel set up, and the data into all of them is bit sync'd from (say) a six channel source, the outputs will be bit sync'd. If the inputs are not quite bit aligned, then the outputs will not quite be bit aligned. If the inputs are way off in phase but frequency locked, the outputs will be way off in phase but frequency locked, and the timing of the block structures of the several output signals will not be defined.

Sample Rate (kS/s)	Time in input sample rate UI's[20]	Time in ns, 44.1 kS/s input	Time in ns, 96 kS/s input
Basic ***dCS 974*** unit to unit output timing alignment.	± 0.3	± 50	± 24
Input misalignment allowed, with all inputs at the same frequency, for block structuring to work (no Master Clock).	± 2	± 334	± 162
Input misalignment allowed, with all inputs at the same frequency, with Master Clock	± 4	± 668	± 324
Input misalignment allowed, with all inputs at the same frequency, with Master Clock as in **Figure 16**[21].	± 64	± 11338	± 5208

Table 8 – Multiple Channel Sync mode – allowable input misalignment

The scope shots below show the timing relationships between two units, using the sync link, for various conversions. They are taken from 2 units, with scope probes on the same point in each unit. The units are linked by a sync link.

[19] version 2.0 software or higher.
[20] UI = Unit Interval, see AES3 spec. There are 128 UI's per sample in AES3.
[21] Master Clock can adjust its Wordclock phases as necessary.

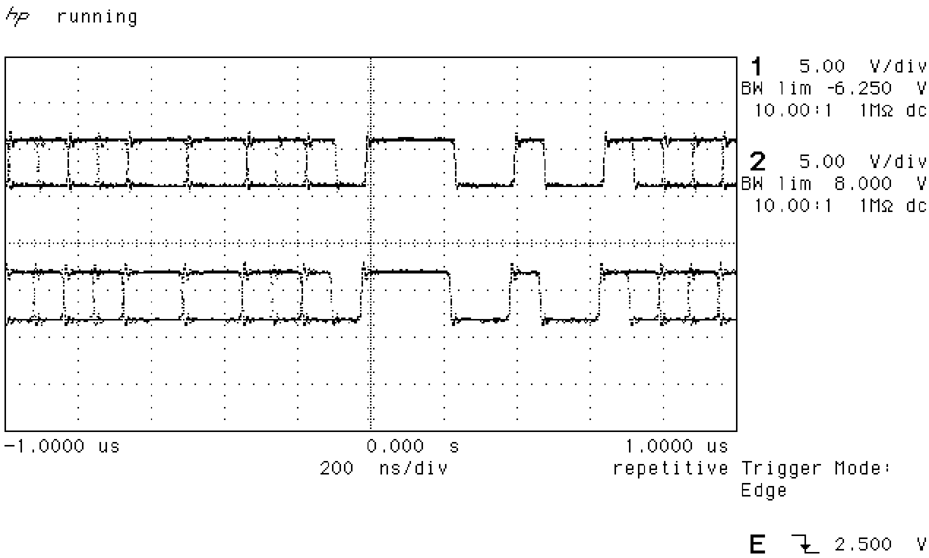


Figure 28 – 44.1 kS/s to 96 kS/s PCM, showing AES3 alignment

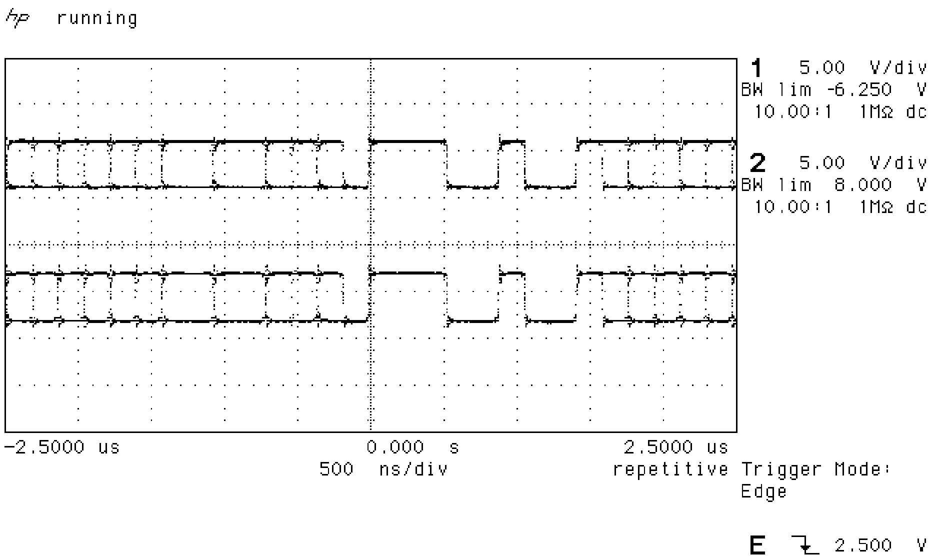


Figure 29 – 96 kS/s to 44.1 kS/s PCM, showing AES3 alignment

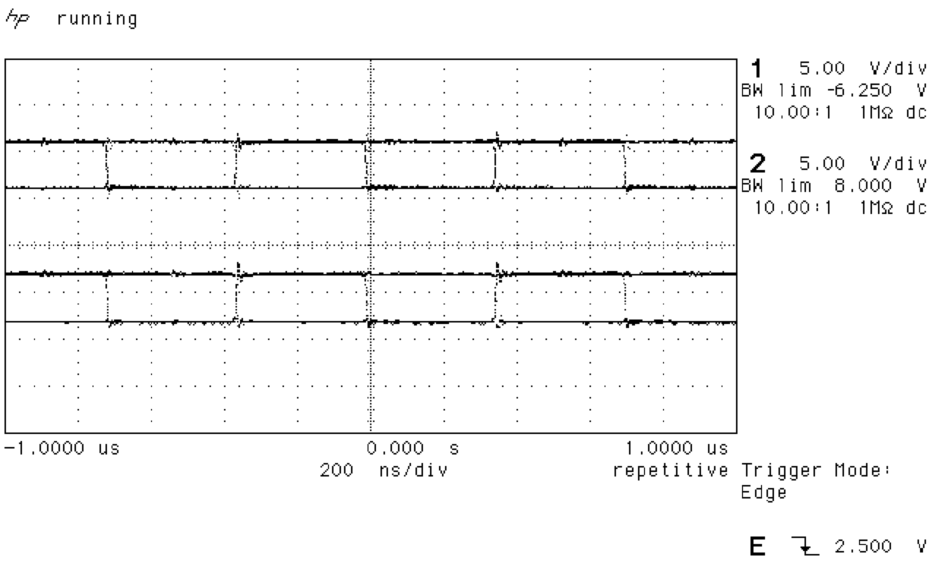


Figure 30 – 44.1 kS/s to DSD, showing DSD bit alignment

Multiple Channel Multiple Sample Rate Synchronising

DVDs can make use of mixed sample rates in their multi-channel mode – for example, 96 kS/s in the front channels and 48 kS/s in the rear channels.

As well as the problems mentioned above in the section **Multiple Channel Sync'ing**, use of multiple channels brings an additional problem in that the group delay in the sample rate conversion process varies with the sample rate. It generally gets longer as the sample rate gets lower, making the rear channels seem further away than you might want.

Figure 16 shows how a master clock may be used with several ***dCS 974***'s in parallel to get simultaneous synchronised sample rates out at the same time.

Please contact ***dCS*** if you need support for this process.

Noise Shaping

The ***dCS 974*** uses noise shaping[22] that is optimised to the F weighting curve[23]. It does not affect signal frequency or transient response, but shapes the frequency response of errors (Q noise, or truncation errors) so that they fall as much as possible in the less sensitive part of the spectrum. The architecture used also shapes dither, where this is added. For all the major sample rates (32 kS/s, 44.1 kS/s, 48 kS/s, 88.2 kS/s, 96 kS/s) the noise shapers have been individually optimised and the first 10 orders are offered. The 1st, 3rd, and 9th shapes for 44.1 kS/s agree well with Wannamaker's published results[24].

Noise Shaping adds more noise power, but because of the shaping it is perceived as lower noise. There is a compromise to be drawn – as more aggressive shaping is used, more noise is added, and less perceived improvement occurs. In practice, things stop improving by about the 10th order. The increased real noise power can cause (small) clicks in editing, if this is carried out after the shaping. For each major sample rate, we recommend an aggressive and a gentle setting, below:

Sample Rate (kS/s)	Lowest Order	Highest Order	Gentle Shaping	Aggressive Shaping
32	1st	10th	2nd	6th
44.1	1st	10th	2nd	9th
48	1st	10th	2nd	8th
88.2	1st	10th	3rd	7th
96	1st	10th	3rd	7th

Table 9 – Noise Shape Orders by Output Sample Rate

The gentle shaping tends to follow the E weighting curve, by chance. For more information on this topic, either see **Word Length Reduction** on page **104** or read the references below.

[22] It actually uses an Error Shaping architecture, but the name is now being used for entirely other things and is less well known, so we call it, erroneously, Noise Shaping

[23] "Minimally Audible Noise Shaping", S.P.Lipshitz and R.A.Wannamaker, J AES vol 39 no 11, p836-852

[24] "Psychacoustically Optimal Noise Shaping", R.A.Wannamaker, J AES vol 40 no 7/8, p611-620

Dither

The ***dCS 974*** uses relatively unusual dither generators to achieve the very good statistics necessary for audio purposes. Many dither generators use PRBS generators (Pseudo Random Binary Shift register generators). These are well known, well documented, and very predictable – but unfortunately their statistics are not that great. The problem shows up as a spectrum that is not flat, and histograms (PDFs, Probability Density Functions) that diverge from the ideal quite significantly.

Adding dither adds noise. Top hat dither uses one generator per channel to add ±0.5 lsbs p-p of rectangularly distributed dither, and triangular dither uses two generators per channel to add ±1 lsbs p-p of triangularly distributed dither. This is on top of the Q/√12 rms dither from word length reduction in the first place (where Q is the size of the output word lsb). The noise shaped triangular setting uses one generator to add ±1 lsbs p-p of triangularly distributed dither that is frequency shaped, and so has low perceived (weighted) noise power. This last is a significant test of generator performance – the ***dCS 974*** performs very well. Performance curves for all these settings, including PDFs of the added dither, are given in the **dCS 974 Performance Curves,** page **94**.

The noise added by the dither settings is summarised below:

Dither Type	Noise Power (0-Fs/2)	Added Noise Power (unweighted)	Perceived Noise Added (F weighted)
No dither (straight truncation)	Q/√12	0 dB	0 dB
Top Hat dither	Q/√6	3 dB	3 dB
Triangular dither	Q/√4	4.8 dB	4.8 dB
Noise Shaped Triangular dither	Q/√4	4.8 dB	1.2 dB

Table 10 – Noise Added by Dither Types

If dither is used, it is shaped (made less audible) by noise shaping. This applies as much to noise shaped triangular as the others.

Digital Interface Specifications

AES/EBU (AES3)		Input	Output	
Type		*Balanced, differential*		
Impedance		110	110	Ω
Sensitivity (unloaded)		1 ~ 10	7	V pk-pk
Maximum Wordlength		24	24	bits
Connector		XLR3 female	XLR3 male	
Connections	Pin 1	Ground or shield		
	Pin 2	+Signal		
	Pin 3	-Signal		

Table 11 – AES/EBU Interface Electrical Characteristics.

SDIF and DSD	Input (Upper block)	Output (Lower block)	
Type	*Single ended, ground referred*		
Impedance	100 (CH1/2) ~1k (WCLK)	55	Ω
Sensitivity (unloaded)	TTL	TTL	
Maximum Wordlength	24	24	bits
Connector	BNC x 4	BNC x 3	
Connections	CH1 IN	CH1 OUT	
	CH2 IN	CH2 OUT	
	WCLK IN / LOOP OUT	WCLK OUT	

Table 12 – DSD/SDIF Interface Electrical Characteristics.

IMPORTANT!

The upper DSD/SDIF **WCLK IN** *connector loops through to the upper* **LOOP OUT** *connector to allow an external Wordclock to be daisy-chained. The last unit in the daisy-chain will need a 75Ω BNC termination connected to the upper* **LOOP OUT**.

SPDIF (electrical)	Input	Output	
Type	*Single ended, ground referred*		
Impedance	75	75	Ω
Sensitivity (unloaded)	0.5	1.0	V pk-pk
Maximum Wordlength	24	24	bits
Connector	RCA Phono & BNC		

Table 13 – SPDIF Interface Electrical Characteristics.

SPDIF (optical)	Input	Output	
Type	*Optical*		
Maximum Wordlength	24	24	bits
Wavelength	660nm		
Connector	Toslink EIAJ CP-340		

Table 14 – SPDIF Optical Interface Characteristics.

IMPORTANT!

*The Toslink interface is not specified for operation at 88.2 or 96 kS/s due to limitations in the Toslink devices. In practice, the interface has been shown to operate correctly with other **dCS** equipment at all sample rates up to 96kS/s but **dCS** cannot guarantee this due to Toslink manufacturing variations.*

Message Handling

The ***dCS 974*** strips the message information from the incoming data. The message in the outgoing data is set using the AES Message Edit and SPDIF Message Edit menus, located under the Sample Rate Conversion menu.

AES/EBU Message Handling

The AES3-1992 standard was written at a time when only 3 sample rates were in common use in the audio industry. ***dCS*** started manufacture of ADCs featuring 96 and 88.2 kS/s sample rates in 1993 and we have evolved modifications to AES3-1992 which accommodate these innovations. These are currently being considered by the AES.

The AES/EBU interface transmits a data structure that conforms to the ***dCS*** version of AES3-1992. This contains 28 bits of Manchester encoded data, and a 4 bit near-Manchester encoded preamble in a subframe, and subframes are further assembled in a block and frame structure. Each subframe contains:

- preambles, to allow the receiver to sync up
- up to 24 bits of audio data, transmitted lsb first
- V, a validity bit
- U, a user bit, for the "User Message"
- C, a Channel Status bit, for the "System Message"
- P, a parity bit

IMPORTANT!

The AES/EBU interface and the SPDIF interface have similar data structures, although the messages are completely different. The two structures are identified in the data domain by the use of the Consumer/Professional bit (bit 1 in the message). A "1" indicates AES/EBU format, a "0" indicates SPDIF format.

The default AES/EBU message attached to the output data by the unit before being changed by the user is as follows:

Professional:	On
Non-Audio:	Off
Mode:	Stereophonic
Source:	DCS1
Destination:	null

For more information on the way ***dCS*** implement the AES3 system message to handle higher sample rates, see the Appendix to this manual. For the formal definition of the AES3 interface, see footnote[25], from the AES.

[25] AES3-1992 (ANSI S4.40-1992) “AES Recommended practice for digital audio engineering – Serial transmission format for two-channel linearly represented digital audio data”.

SPDIF Message Handling

The SPDIF interface (sometimes known as the Consumer AES/EBU interface) transmits a data structure that conforms to the IEC 958[26] standard. Like the AES/EBU, this contains 28 bits of Manchester encoded data, and a 4 bit near-Manchester encoded preamble in a subframe, and subframes are further assembled in a block and frame structure. The difference lies only in the voltage levels, and the Channel Status bits (the System Message). It contains 24 bits of audio data.

IMPORTANT!

The AES/EBU interface and the SPDIF interface have similar data structures, although the messages are completely different. The two structures are identified in the data domain by the use of the Consumer/Professional bit (bit 1 in the message). A "1" indicates AES/EBU format, a "0" indicates SPDIF format.

The default SPDIF message attached to the output data by the unit before being changed by the user is as follows:

Professional:	Off
Non-Audio:	Off
Copy Permit:	On
Format:	2-Channel General Format

[26] See EN 60958:1995 or IEC 958:1989 with amendments 1&2. The structure of the message is sufficiently complex that it is best to read the source material.

SDIF-2 Message Handling

The SDIF-2 interface is a 4 wire NRZ interface - so the DC level on each signal line may not be constant. It contains 20 bits of audio data and has a block structure of 256 stereo samples, rather than the 192 of AES/EBU. There are 8 bits of message per channel per sample - with a further 3 bits being used for an "illegal code" based sync code. Of the 8 bits per sample, the 8 in the first sample are reserved for system messaging, and the rest are for User messages.

The 4 wires are:

Ground return
Left Channel
Right Channel
Wordclock

The sync codes enable data recovery without the word clock, if necessary, but with the number of data formats in current operation, this method of locking is strongly discouraged.

The SDIF-2 message is given in the table following. The ***dCS 974*** implementation sets all bits of the User message to "0".

DESCRIPTION	Definition	Default Message
Undefined	0000 0xxx	0000 0xxx
Emphasis		
No emphasis	xxxx x00x	xxxx x00x
Emphasis (15µsec, 50µsec)	xxxx x01x	
Dubbing Prohibit		
Dubbing allowed	xxxx xxx0	xxxx xxx0
Dubbing inhibited	xxxx xxx1	
Block Code		
Start of block	xxxx xxxx 1...	as required
Not start of block	xxxx xxxx 0...	as required

Table 15 - SDIF-2 Message Table

SDIF-3 Message Handling

At present, no messaging is implemented in SDIF-3.

Power Consumption

The ***dCS 974*** has a linear power supply, and so power consumption changes as the supply voltage changes. The internal regulation is comparatively efficient for a linear supply, so these changes are kept to a minimum. Consumption is independent of supply voltage setting.

Power Consumption with Supply Voltage (measured as AC power into mains socket):

Nominal voltage	16 W
Voltage -10%	14 W
Voltage +10%	17 W

The actual intended supply voltages are shown on the rear panel. 50Hz or 60Hz operation is not important – the unit can use either. In general, users will not need to change the mains input configuration. If you do need this to be done, please see the section **Having Your Options Changed,** page **114** in this manual and contact your distributor or ***dCS***.

Size, Weight and Operating Conditions

Size and Weight

The ***dCS 974*** dimensions correspond to a standard 2U 19" rack mount case. Four heavy duty feet, fitted to the base, extend the overall height to slightly greater than 2U.

Dimensions

Width	430 mm	see note (i)
Height, without feet	88 mm	(2U)
Height, with feet	95 mm	
Depth	390 mm	see note (ii)
Weight	8.5kg	see note (iii)

note (i)	Removable 19" rack mount ears are supplied, taking total width to 483 mm (19").
note (ii)	Measured from front panel to rear panel connectors. Additional depth should be allowed to accommodate cable connectors. The rotary control knob protrudes 22mm from the front panel.
note (iii)	The high quality case is necessarily heavy, consideration should be paid to appropriate support shelving when installing the units in a rack.

Operating Conditions

The case of the ***dCS 974*** has no ventilation slots or fan cooling, to give:

- quiet operation (does not need to be installed in a machine room)
- internal temperature stability
- improved electrical safety
- long term reliability
- no regular maintenance or cleaning requirements

It dissipates relatively low power, so that usually allowing natural convection provides enough cooling. Do not install the unit near heat sources such as radiators, air ducts or direct strong sunlight. Ambient should not exceed 50°C, should not fall below 0°C, and should be non condensing. If in doubt, the easy test is – the ***dCS 974*** is happy to work anywhere a human is.

dCS 974 PERFORMANCE CURVES

The graphs on the following pages show key performances of the ***dCS 974***.[27]

- Dither PDFs, showing how the dither types are effectively ideal, including noise shaped triangular dither
- FFTs of the dither signals used (excludes truncation, or Q, noise)
- Raw data for small signal truncated (± 4 lsbs) sinewave, showing source waveform, no dither, and all dither types. Shows absence of DC offset, and behaviour of the various dithers
- FFTs of truncated small signal (-90 dB0 sine), showing no truncation, truncation without dither, and the effect of dithers.
- FFTs of truncated small signal (-90 dB0 sine), with noise shaping, with and without added dithers.
- DSD spot Q noise for filter options
- DSD integrated Q noise for filter options
- DSD F weighted Q noise for filter options
- DSD 0 dB0 (full scale) signal
- 48 kS/s to 44.1 kS/s conversion, Filter responses
- 96 kS/s to 44.1 kS/s conversion, Filter responses
- Noise shaping curves for 32 kS/s
- Noise shaping curves for 44.1 kS/s
- Noise shaping curves for 48 kS/s
- Noise shaping curves for 88.2 kS/s
- Noise shaping curves for 96 kS/s
- Noise Shaper noise and weighted noise vs shaper order

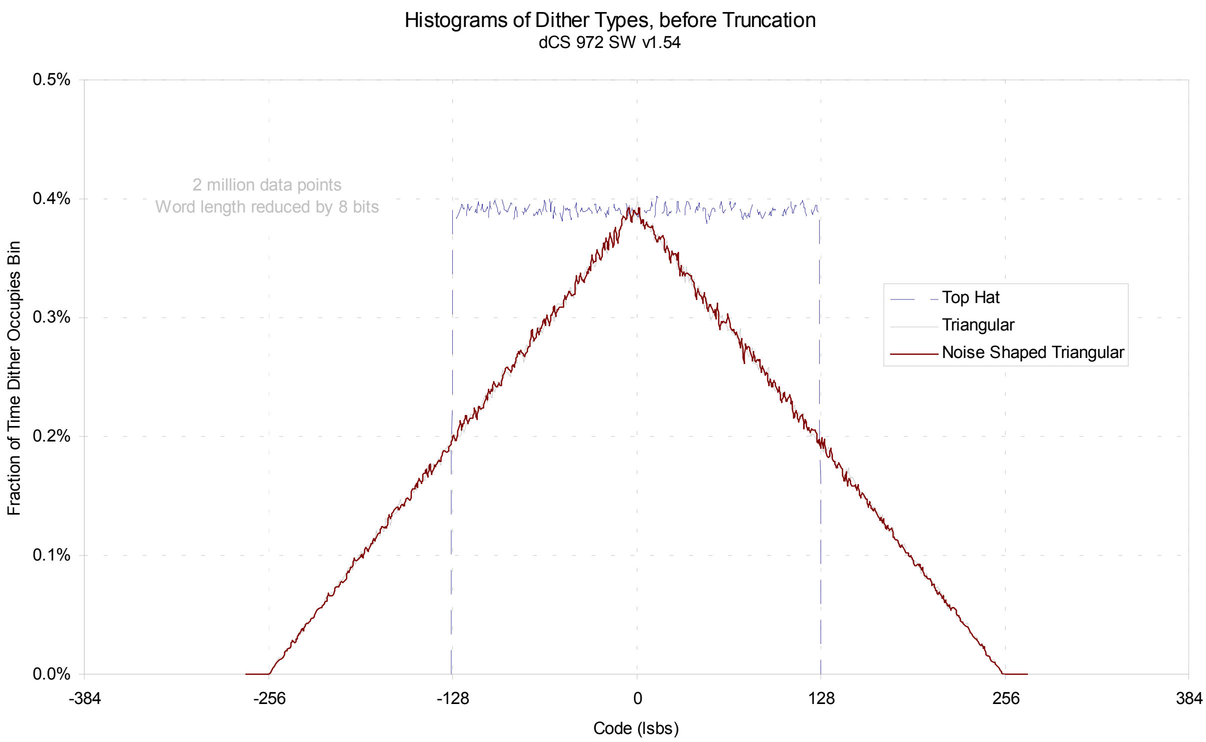


Figure 31 – Dither PDFs

[27] Note that many aspects of the ***dCS 974*** are identical to those of the ***dCS 972***.

Figure 32 – Dither Spectra (FFTs)

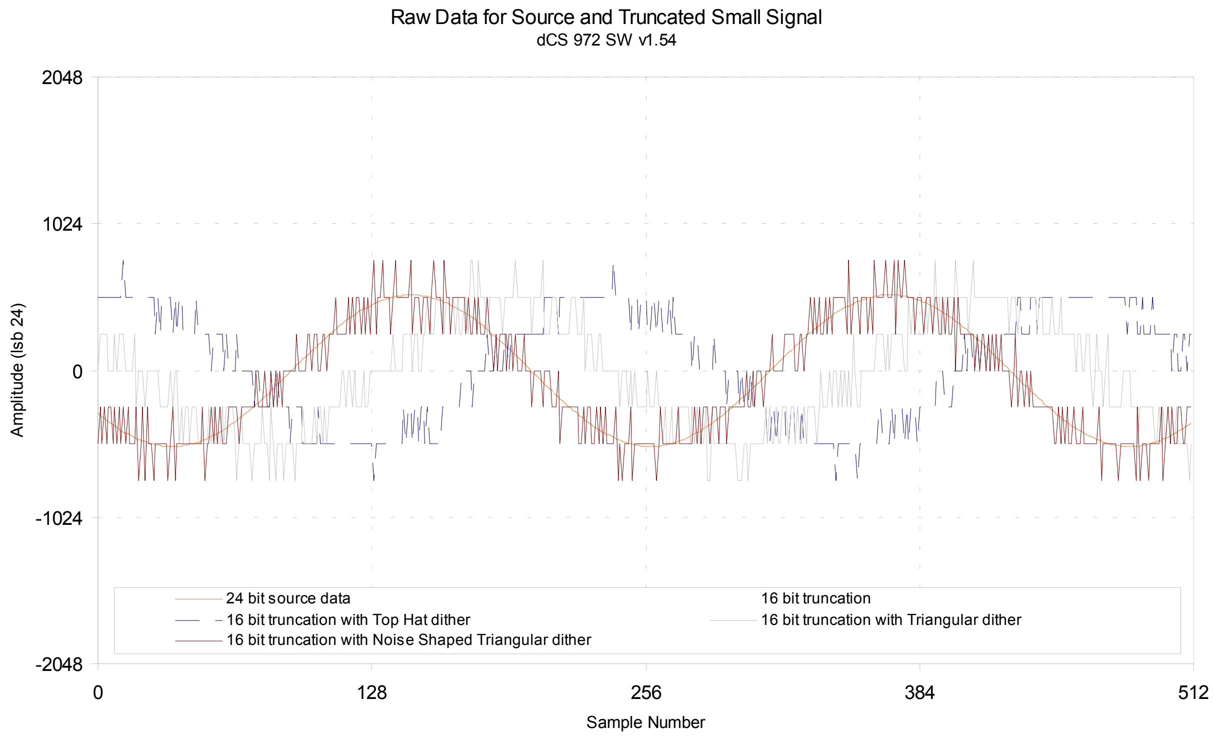


Figure 33 – Small Signal Raw Data with Dithers

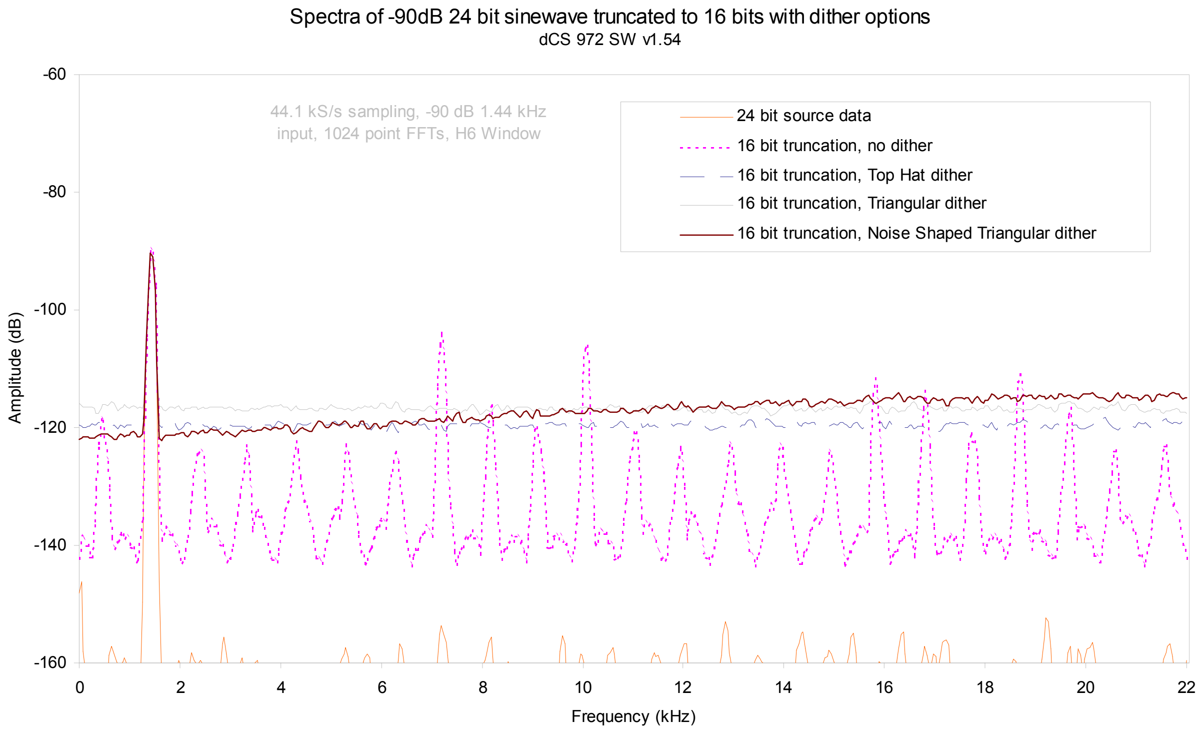


Figure 34 – Truncated small signal FFT, with and without dither

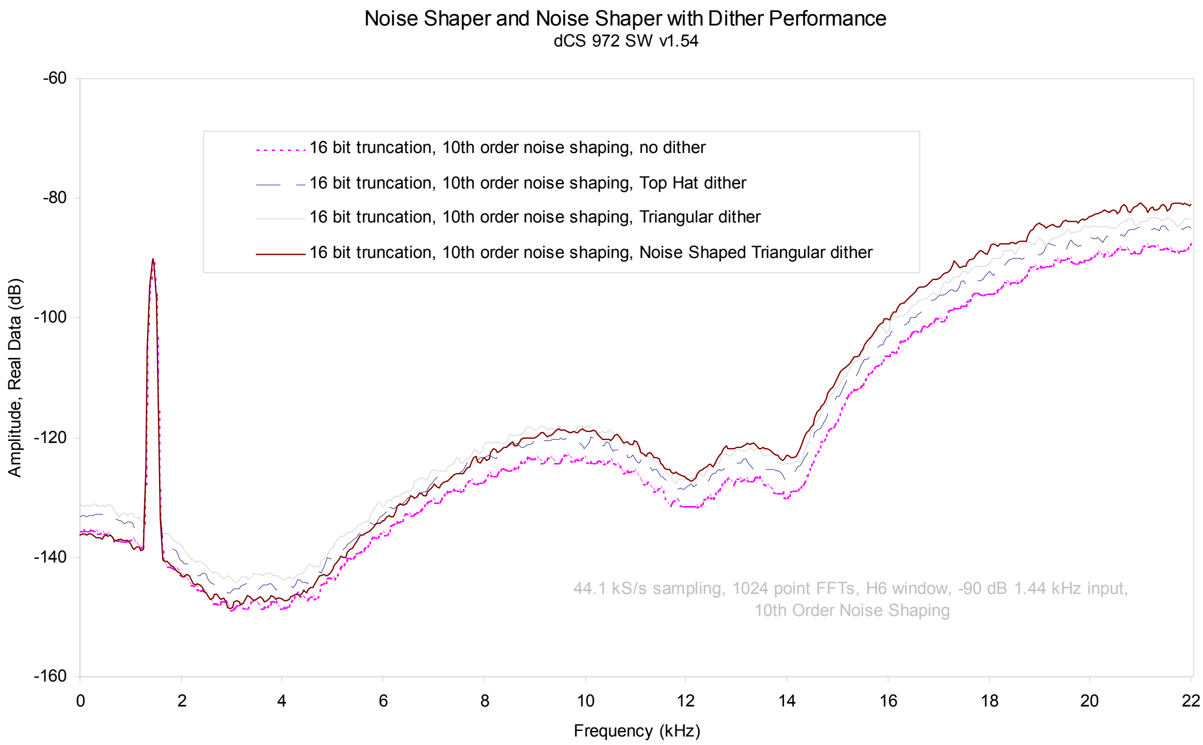


Figure 35 – Truncated small signal FFT, showing noise shaping with and without dither

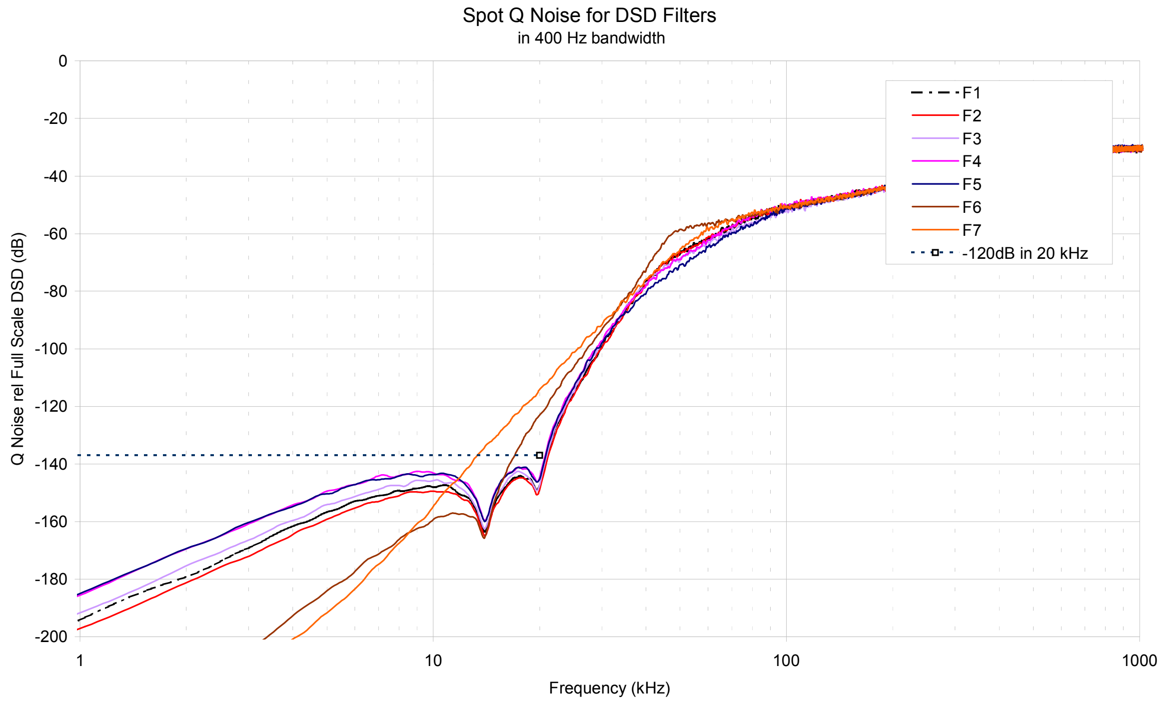


Figure 36 – DSD output, spot Q Noise for the various Filter options

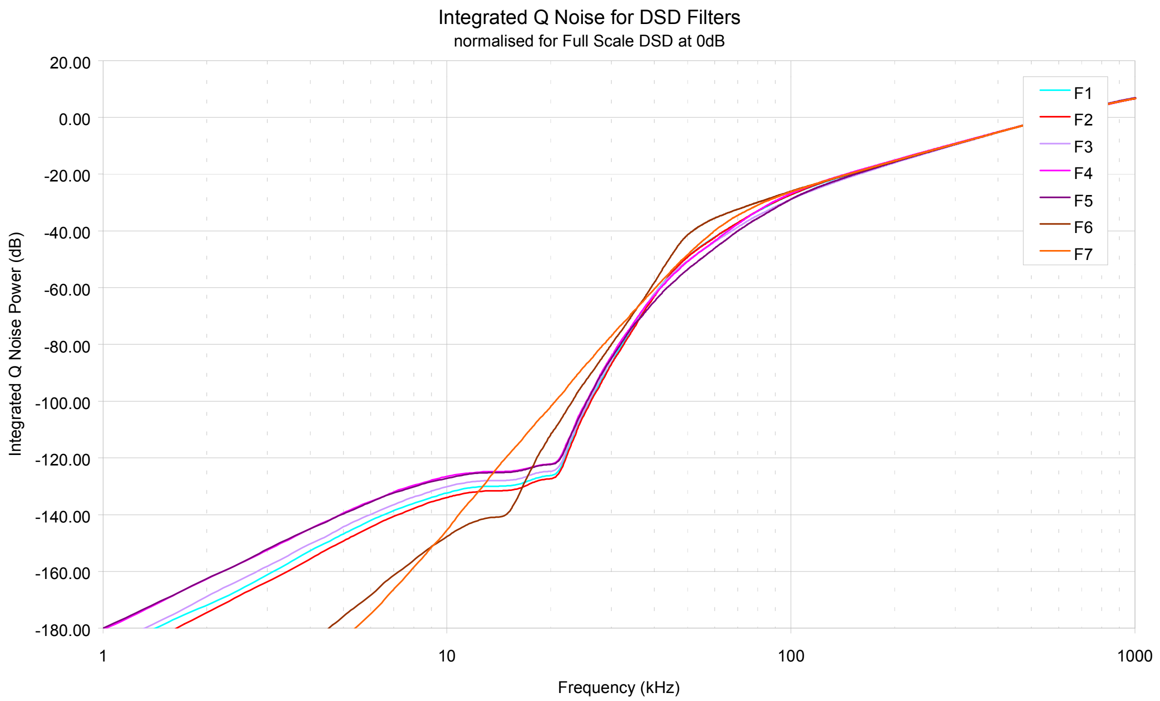


Figure 37 – DSD Output, integrated Q Noise for the various Filter options

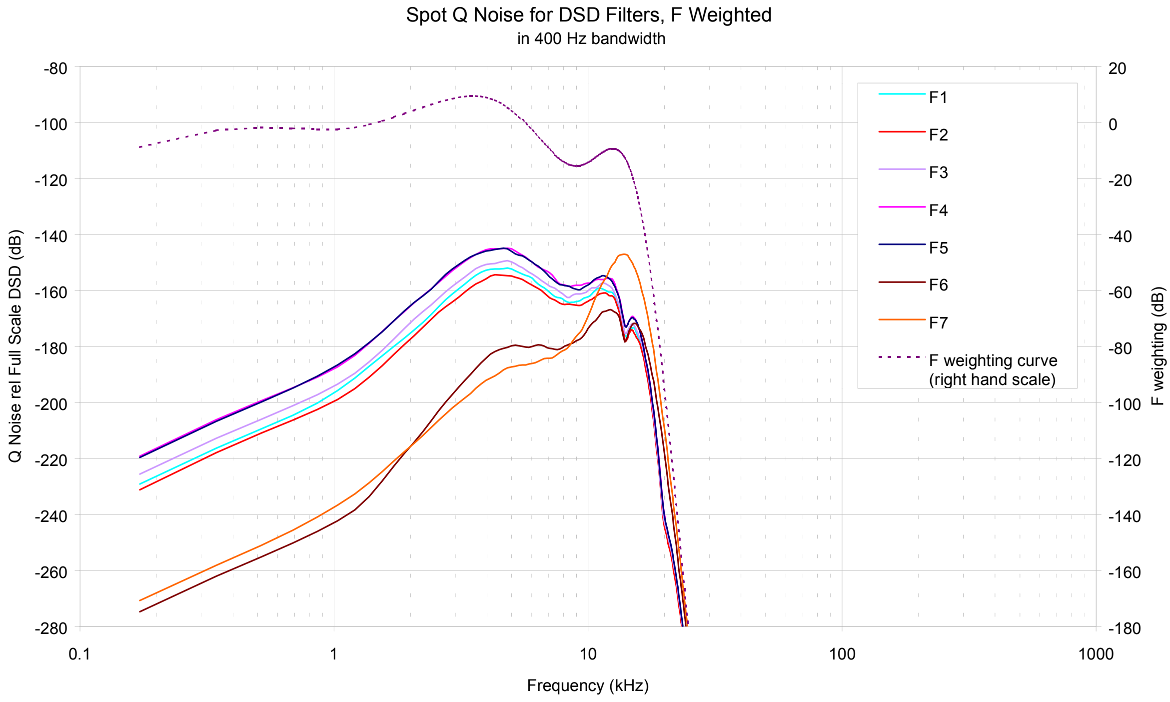


Figure 38 – DSD Output, F weighted Q Noise for the various Filter options

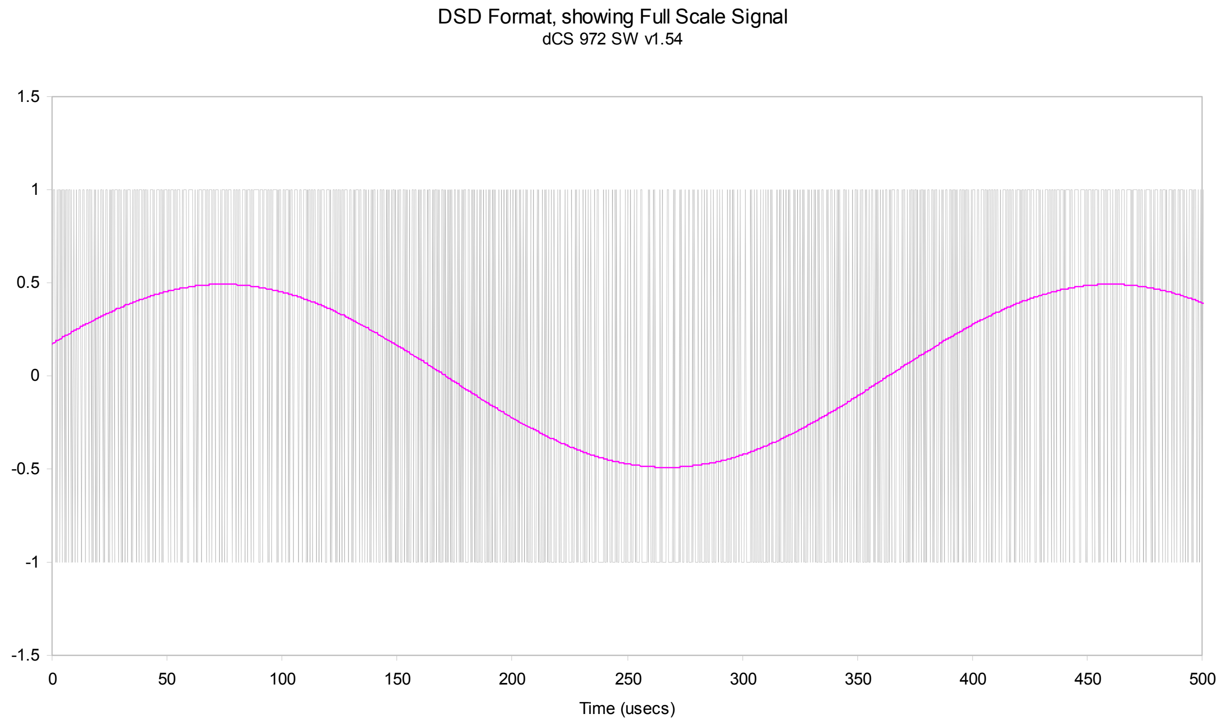


Figure 39 – DSD showing 0 dB0 (full scale)

DSD has only two levels – printer artefacts make it look like more

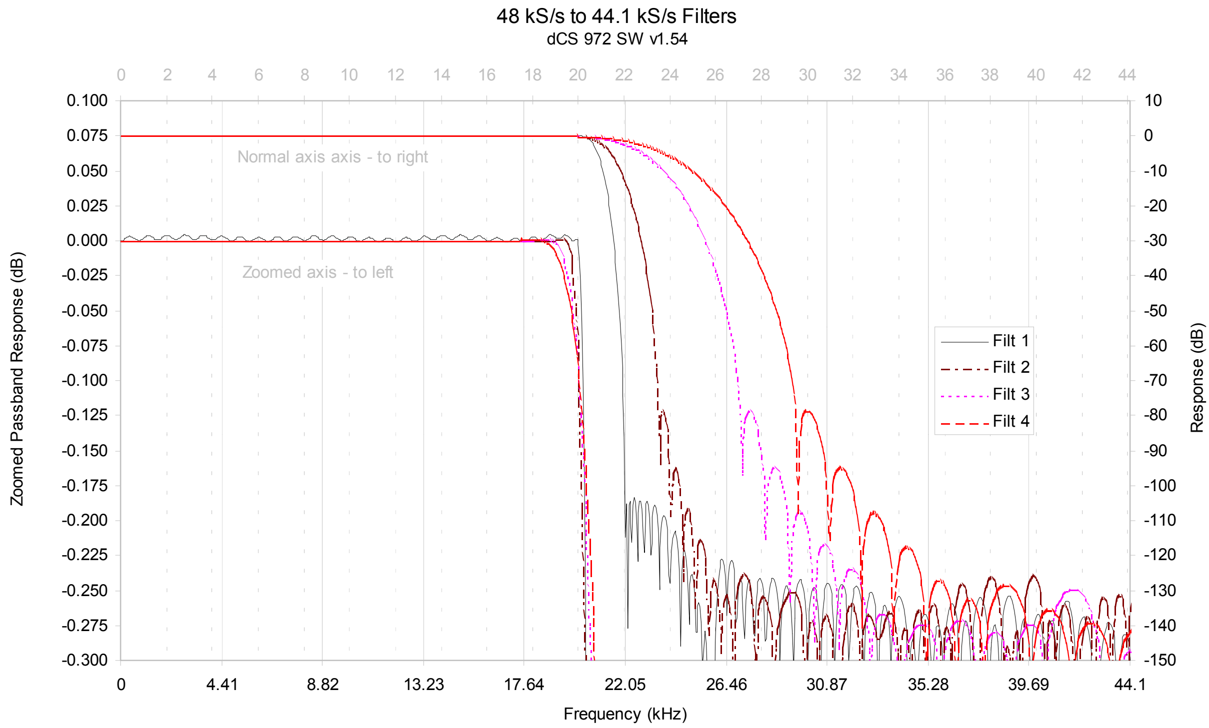


Figure 40 – 48 kS/s to 44.1 kS/s conversion, Filter responses

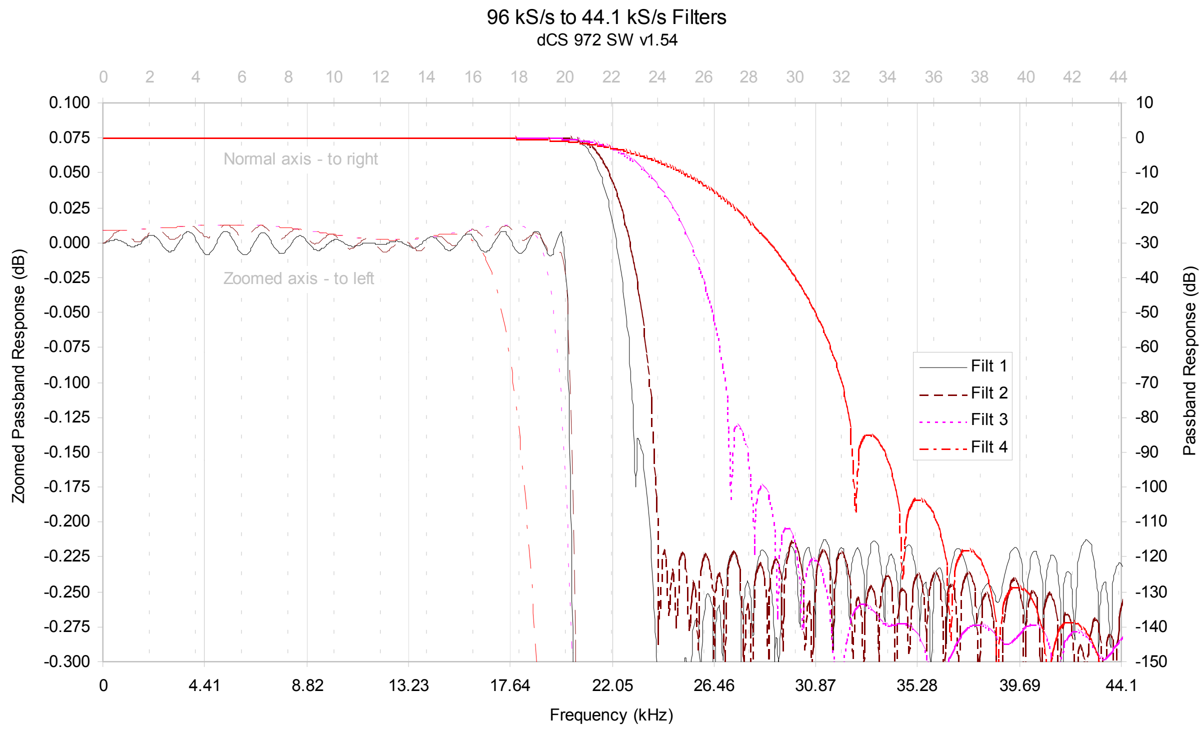


Figure 41 – 96 kS/s to 44.1 kS/s conversion, Filter responses

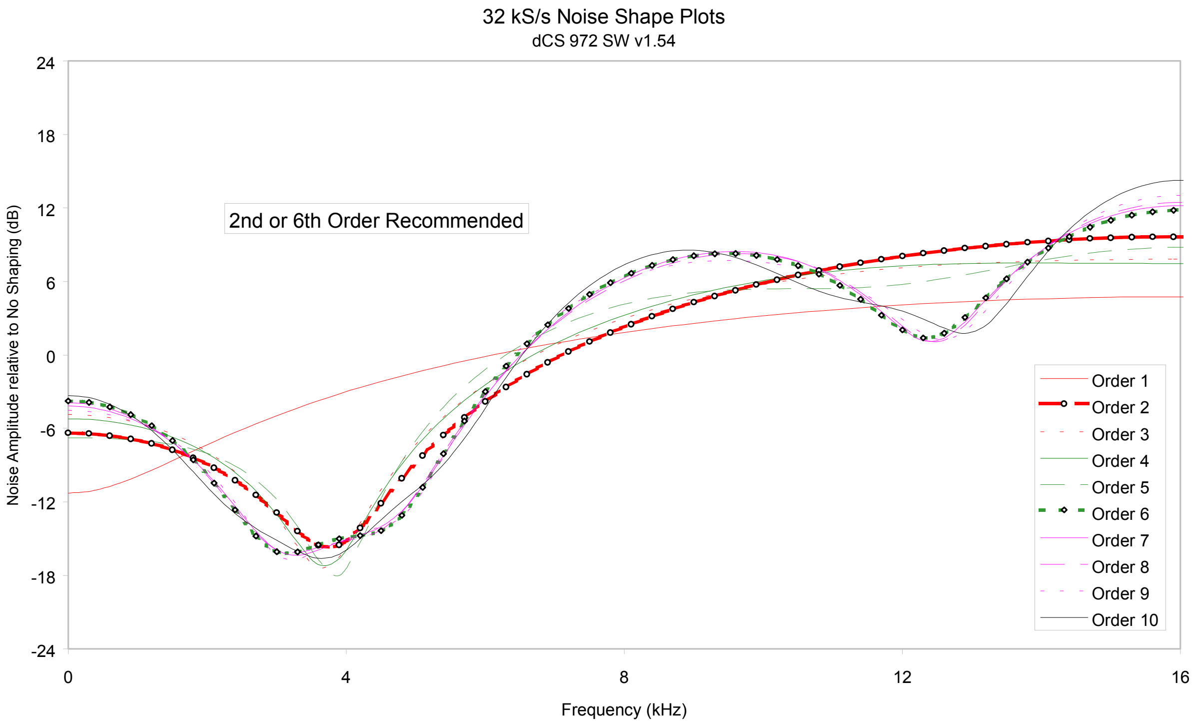


Figure 42 – 32 kS/s noise shaper curves

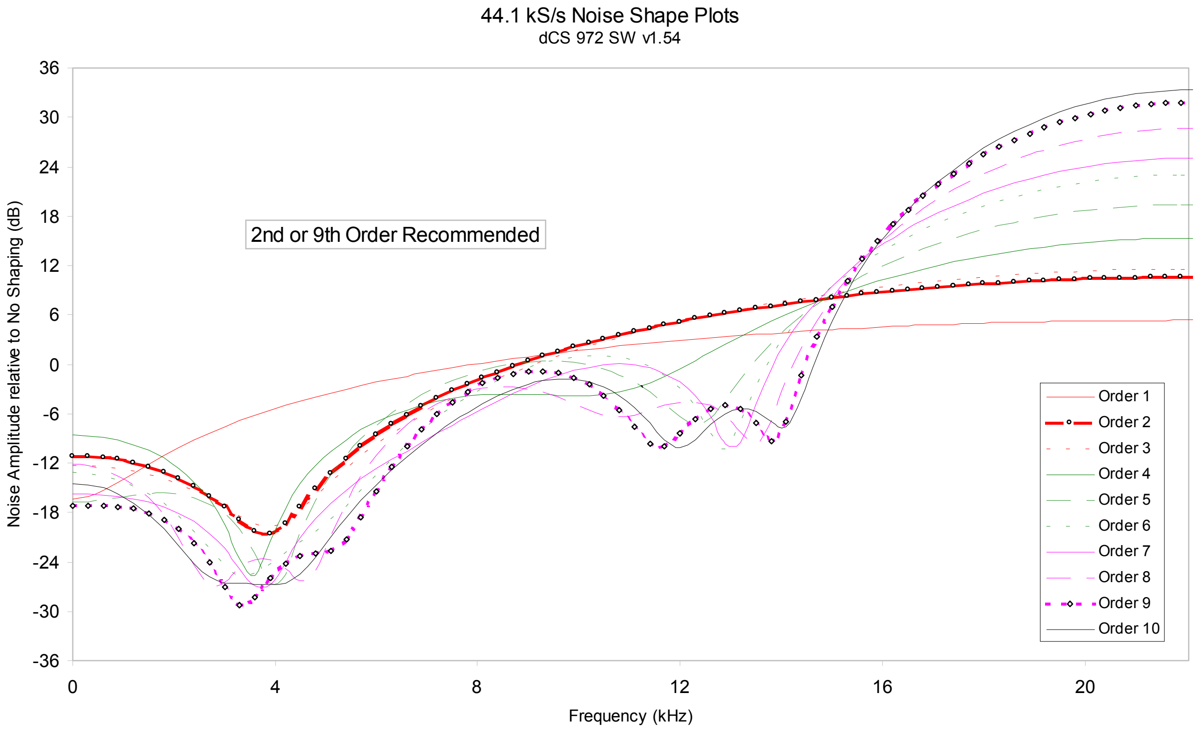


Figure 43 – 44.1 kS/s noise shaper curves

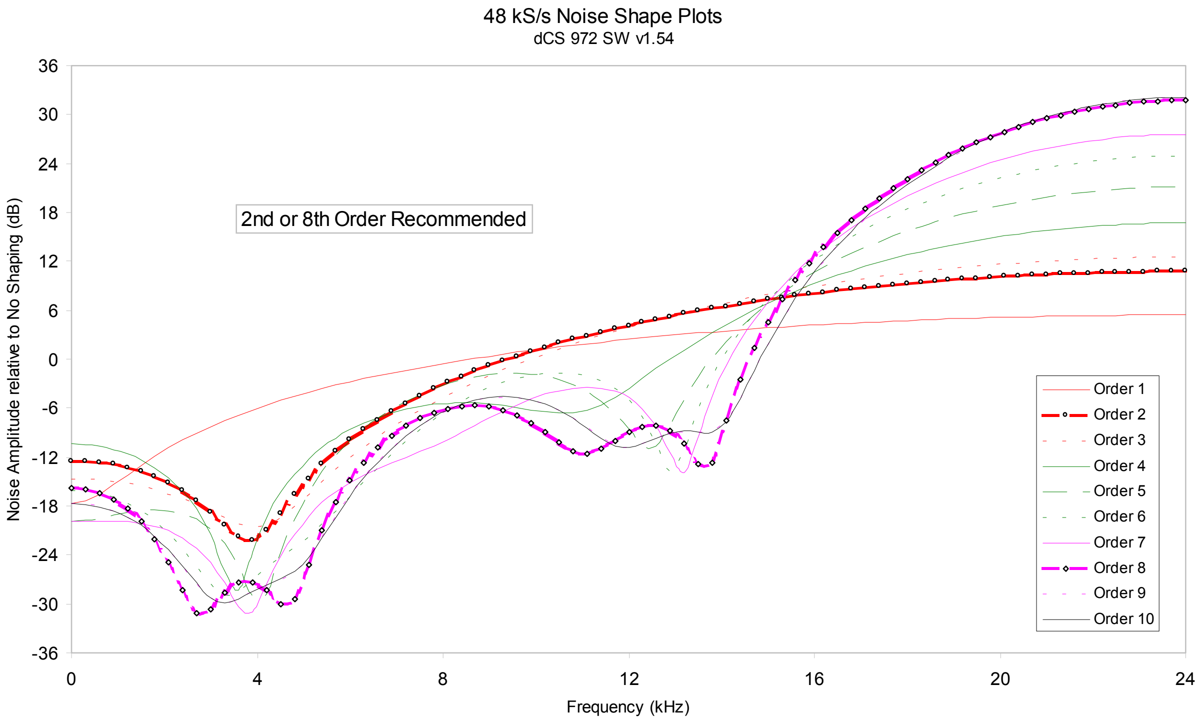


Figure 44 – 48 kS/s noise shaper curves

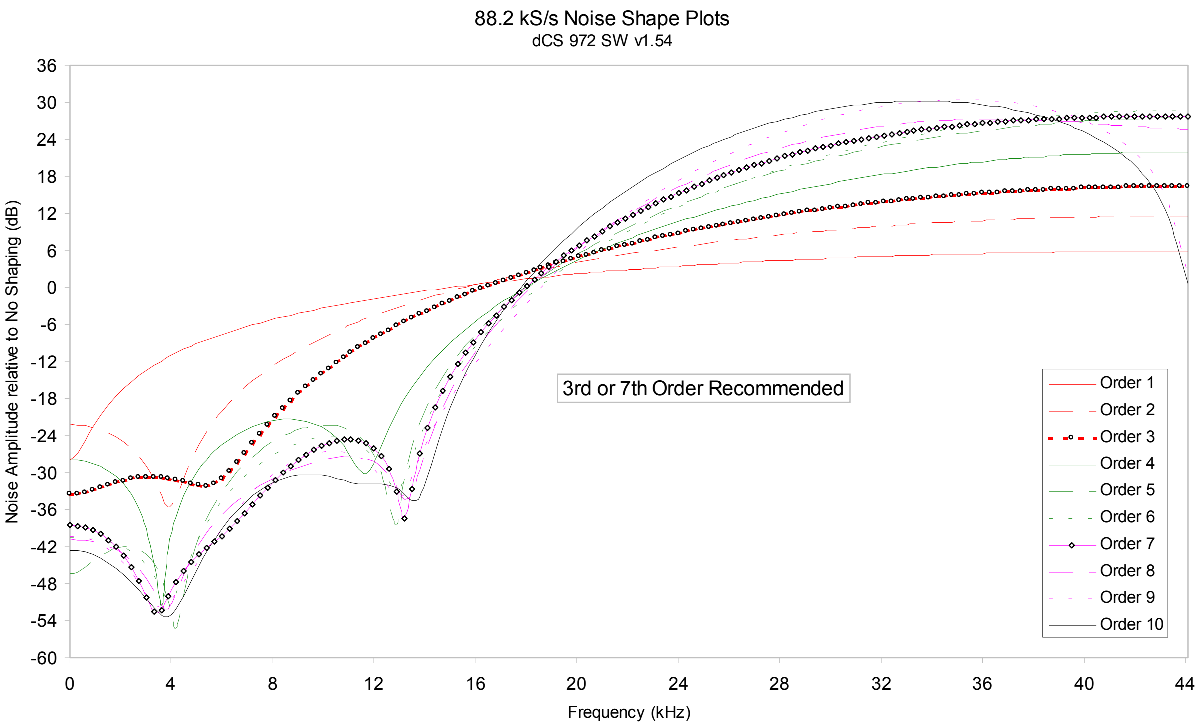


Figure 45 – 88.2 kS/s noise shaper curves

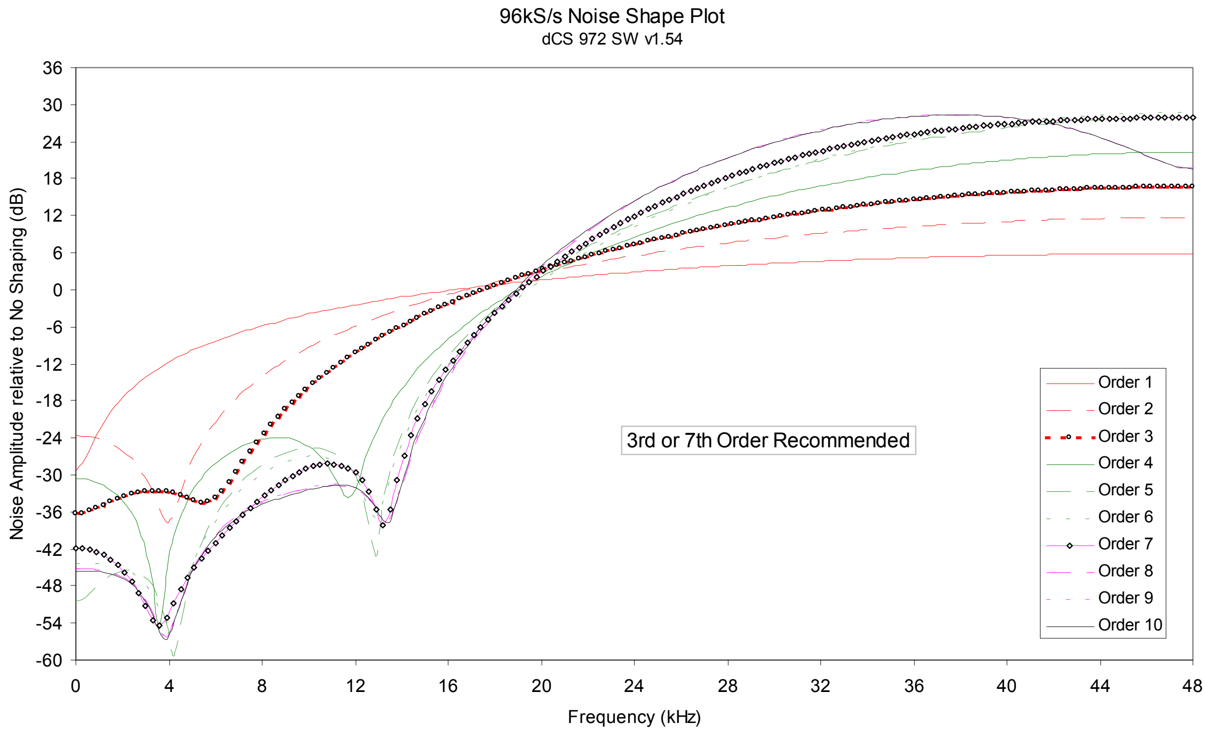


Figure 46 – 96 kS/s noise shaper curves

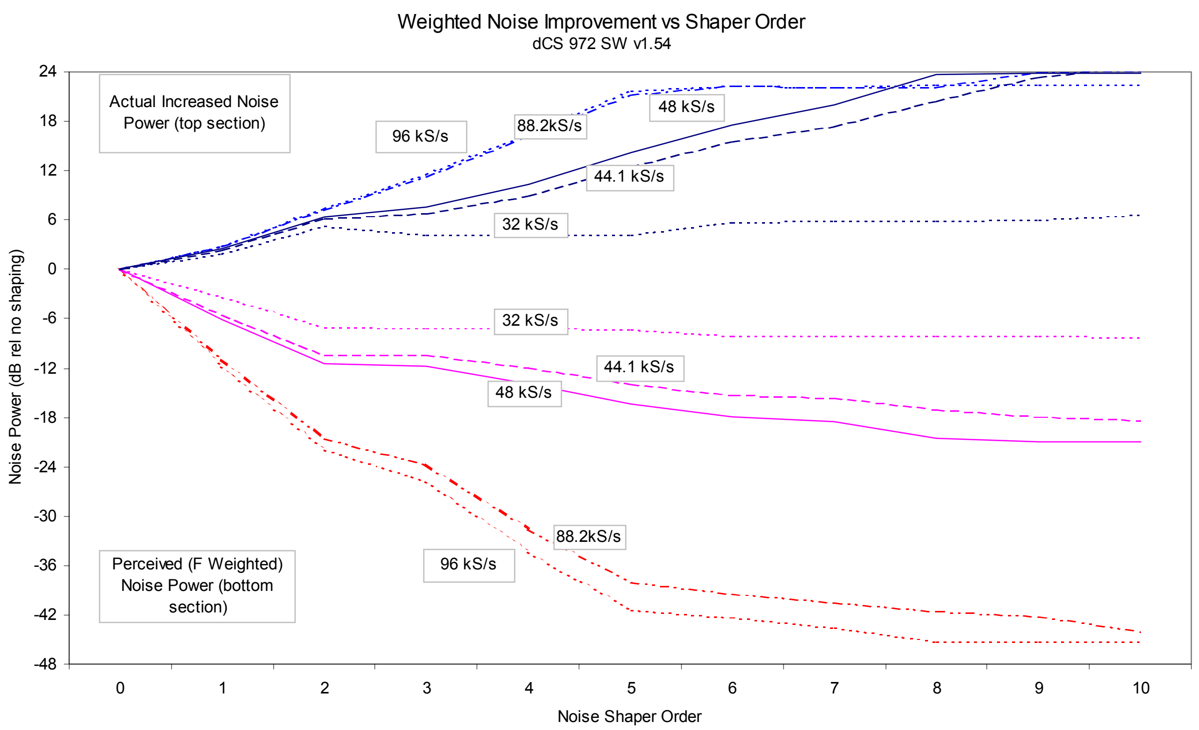


Figure 47 – Noise shaper noise and weighted noise vs order

GENERAL TECHNICAL INFORMATION

Word Length Reduction

Word length reduction (truncation) causes an error signal to be added to the wanted signal. The error signal is usually referred to as “Q noise” or Quantisation noise – the approximation is made that the errors are noise-like. This is true for large signals, but for smaller ones it is not so. As the wanted signal gets smaller, the complexity of the error signal decreases, and the errors first of all pile into ever lower order harmonics or intermods, and then, as the level of the signal sinks below the Q level, much error power piles into the fundamental. This causes its amplitude to become unpredictable – it may drop abruptly to zero and disappear, or it may cease to go down any more and just stay at a constant level. From the audio viewpoint, this sounds very unpleasant. As a signal tail decays away, the tonal quality changes, and then it decays into distorted mush and either abruptly stops, or else keeps fuzzing away until a new signal starts. The level at which all this happens is the lsb of the output word – for CDs, it is at the 16 bit level, which equates to about -90 dB0. The level is high enough to be quite audible, and the effect must be tackled to make reasonable quality CDs.

There is really only one way of tackling the problem – another signal has to be added to the wanted one to smooth the staircase transfer function that truncation causes. Mathematically, with two signals present, the transfer function the wanted signal sees is the convolution of the PDF[28] of the second signal and the staircase function. The converse is also true – the transfer function the additional signal sees is the convolution of the PDF of the wanted signal and the staircase function. This aspect is not a problem with the dither types considered below, but it can be with some highly frequency shaped dithers.

The additional signal is usually referred to as dither, and it is usually noise-like, because then its statistics can be controlled, and the converse effect of the signal modulating the dither can be made insignificant, or zero. However, there are a number of ways that this dither signal can be generated and treated. The major options are:

- it can be generated from the signal or generated independently and added (“Dither”). It seems implausible that the dither signal can be generated from the signal, but it can, and this gives the lowest added noise power option. It is noise shaping on its own, but there are some circumstances where it needs help from additional dither.
- it can be added inside or outside an error shaping loop.
- it can be frequency shaped to match the ears response or not. We can use techniques that suppress error energy in the areas where the ear is sensitive, and put it in areas where the ear is not sensitive. Usually this shuffling around process costs us – we remove a little from the sensitive areas and add back rather more in the less sensitive parts, but that’s life. We still gain some improvements.

The table below gives the actual noise levels for 16 bit truncated signals with no dither, various dither types, noise shaping alone, and noise shaping with dither.

[28] PDF = Probability Distribution Function. References to Rectangular Dither or Triangular Dither refer to the shape of the PDF of the dither.

The 0 dB reference level is taken as the minimum noise we could possibly get away with – the amount that simple 16 bit truncation (16 bit Q noise) would give, if it were well behaved, which it is not.

Straight forward dither always adds noise – it can only produce signals with a noise floor higher than Q noise on its own. However, the noise power added is a few dBs for simple types. Noise shaping adds rather more noise, but it can be made to add it in parts of the spectrum that the ear is less sensitive to, so the perceived noise (F weighted noise) is lower – up to three bits lower. It results in a signal that the ear hears as having a far **lower** noise floor than a 16 bit truncated signal, rather than the "not much worse" of dither alone, even though there is really more noise present[29].

Truncation Type, with 44.1 kS/s data rate	**Noise, unweighted, rel 16 bit Q noise[30]**	**Noise, F weighted, rel 16 bit Q noise**	**Comments**
16 bit truncation	0 dB	0 dB	Unpleasant low level effects
16 bit truncation with Top Hat dither	3 dB	3 dB	Okay – can show noise modulation at low signal levels
16 bit truncation with Triangular dither	4.8 dB	4.8 dB	All noise modulation and unpleasant effects removed, but noise floor is high
16 bit truncation with Noise Shaped Triangular dither	4.8 dB	1.2 dB	All noise modulation and unpleasant effects removed. Not much perceived noise penalty
16 bit truncation with 2nd order noise shaping and no dither	6.2 dB	-10.4 dB	Okay with input noise floors down to –102dB0
16 bit truncation with 2nd order noise shaping and Noise Shaped Triangular dither	11.0 dB	-9.2 dB	Unconditionally free from truncation effects with all inputs
16 bit truncation with 9th order noise shaping and no dither	23.4 dB	-17.9 dB	Okay with input noise floors down to –120dB0
16 bit truncation with 9th order noise shaping and Noise Shaped Triangular dither	28.2 dB	-16.7 dB	Unconditionally free from truncation effects with all inputs

Table 16 – Dither and Noise Shaping Noise Powers

29 DSD carries this further. The principle is the same, but with DSD, there is more noise than there is signal, even at full scale. It is just that it is in a part of the spectrum the ear cannot hear.

30 16 bit Q noise is -98.1 dB relative to a full scale sine wave.

Noise shaping on its own is not perfect. It relies on a small amount of noise in the input signal to generate the frequency shaped correction signal, and if there is very low noise in the input signal, this mechanism can break down. In reality, such a situation can only occur with test signals, digitally generated signals[31], or silences introduced in editing. If one of these situations may arise, any chance of a problem can be completely removed by adding a dither signal as well as using noise shaping. The noise shaping shapes the dither in the ***dCS 974*** architecture. If Noise Shaped Triangular dither is used, then there is very little weighted degradation in the final signal, although the quite high level of total noise power now present means that the process should be carried out after major editing.

There is another option not supported by the ***dCS 974*** – generate the dither independently of the signal and frequency shape it prior to addition, but do not add it in an error shaping loop. This seems to ***dCS*** to combine the worst of all worlds – the high noise floor in the 0 - 6kHz area of straight dither, and the high total noise of noise shaping. However, some people use it.

What does it look like?

Figure 48 gives the spectra of 16 bit truncated 44.1 kS/s signals with a –90dB0 sine present, for two dither only signals (Top Hat, Noise Shaped Triangular), a 10^{th} order noise shaped[32] signal, and a 10^{th} order noise shaped signal with added Noise Shaped Triangular dither. The equivalent simply truncated spectrum is shown in **Figure 49**, separately because it is so revolting. In it, we can see that at the signal level shown (-90dB0) error power from the quantising/truncation is beginning to pile into the fundamental, which is showing an amplitude error of +1.3dB. This would show up on a conventional linearity plot, although the sign of the error could be either way.

31 for example, from synthesisers
32 for comparison with the table, 10^{th} and 9^{th} order noise shaping are very similar.

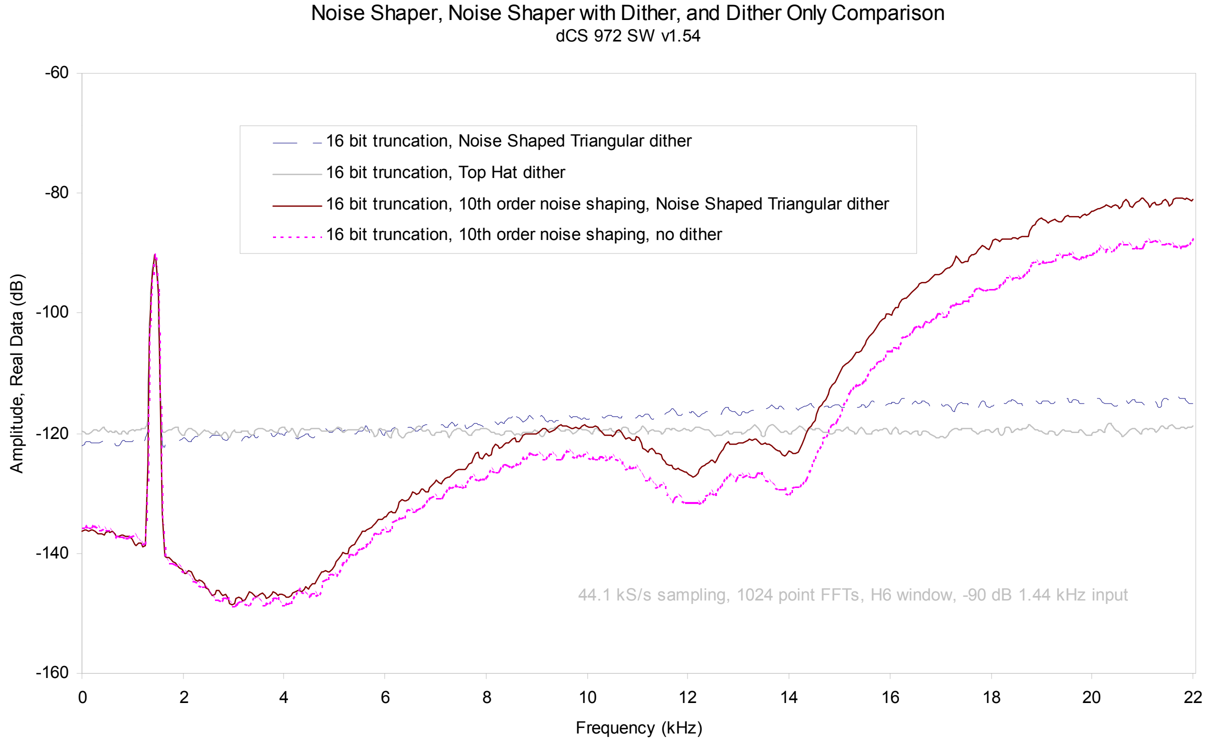


Figure 48 – Noise Shaping and Dither Spectra

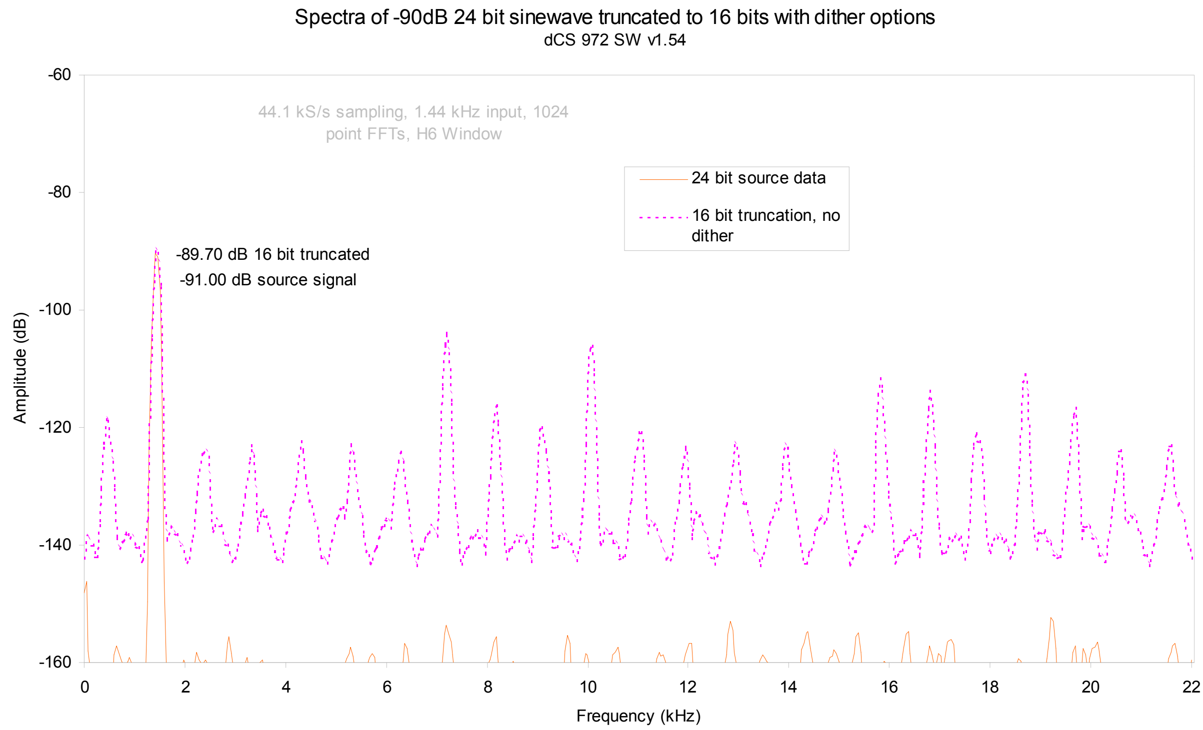


Figure 49 – Truncation Only Spectra

When is Dither Needed?

Dither (on its own or with noise shaping) is not always needed. The condition for not being needed depends on the level of noise in the incoming signal, the amount of noise shaping being used, and the output word length. The user may want to avoid having to think about this at all – in which case, use Noise Shaped Triangular dither – or may wish to minimise noise and added noise.

As a rule of thumb, if the following condition is met, dither is not necessary:

Noise in input signal	+	Noise gain of noise shaper	>	Q noise of wanted output word length

The noise gain of the noise shaper is the same as the unweighted noise power given in **Table 16**, and is shown in **Figure 50** below.

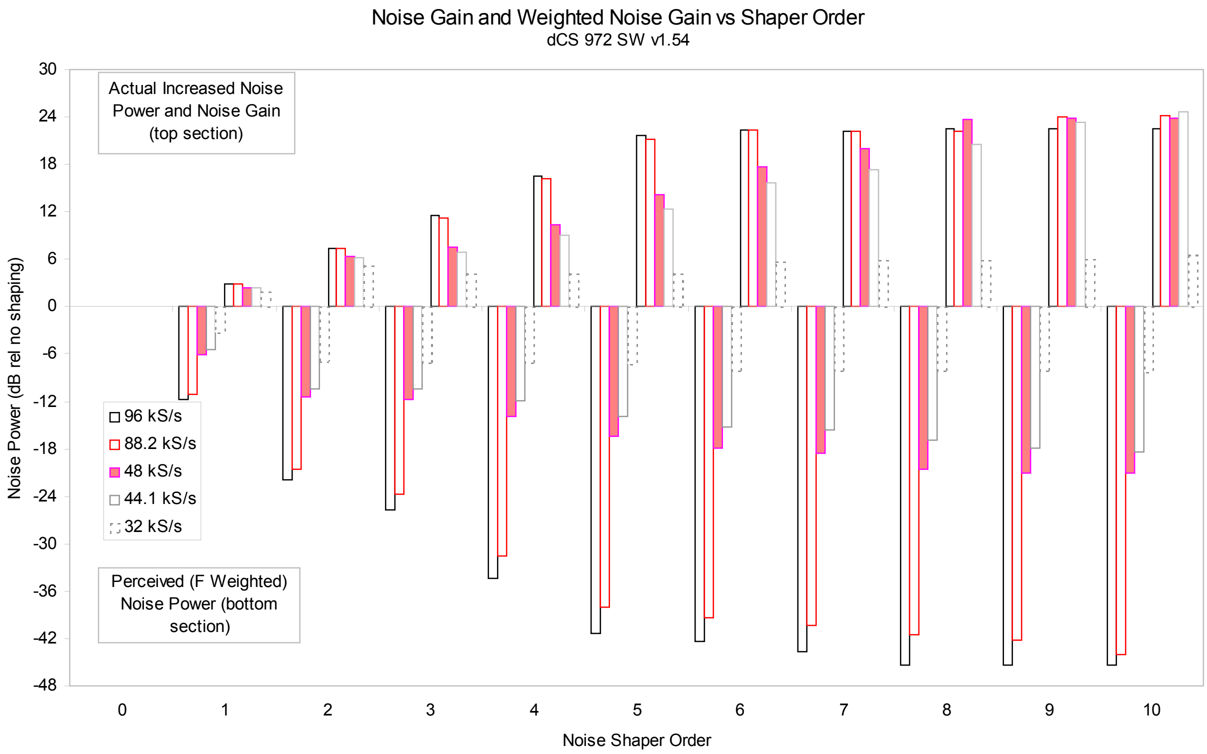


Figure 50 – Noise Gain and Weighted Noise Gain by Shaper Order and Sample Rate

As an example of reading this, consider a 24 bit signal in with a –110 dB0 noise floor, to be output at 48 kS/s 16 bits. The Q noise in the output signal would be -98 dB0, so if a noise shaper with more 12 dB of noise gain was used, no dither would be necessary. Looking at the chart, a 5th order shaper (14.2 dB of noise gain) would avoid the need for dither, whereas a 4th order one (10.9 dB of noise gain) would need dither. The 5th order shaper would give a weighted noise improvement of 16.4 dB, whereas the 4th would only give 13.9 dB perceived improvement, less a further degradation from adding dither.

USING YOUR *dCS 974* FOR THE FIRST TIME

What's in the Box?

The contents of the box are at least:

- ***dCS 974***
- User Manual
- Quick Start Guide
- Power cable
- 2 Spare Fuses
- Remote cable
- Remote software

Supply Voltage Setting

The ***dCS 974*** is shipped with its supply voltage preset for operation in the destination country. The voltage is not intended to be changed by the user. If it needs to be changed, see the section **Having Your Options Changed** on page **114**.

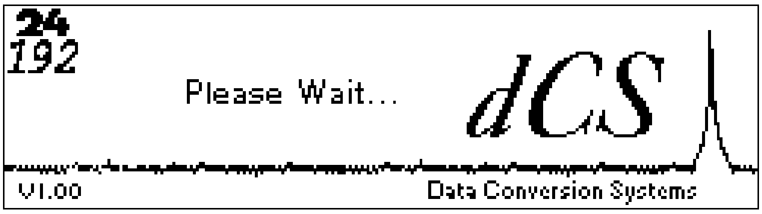
Getting Started

Here's what to do:

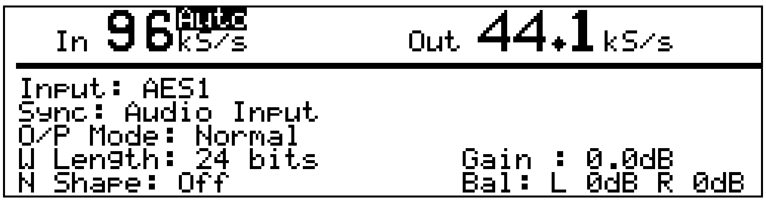
(If the unit does not behave the first time you power up – contact your re-seller or ***dCS***.)

do this: Check the appropriate mains supply for your local mains is marked on the rear panel.

do this: If it is, connect the unit to a power outlet using the cable supplied - connect no other leads at this stage - and switch on. The **Power** LED will light, then the screen will light up and display:



After approximately 20 seconds, the front panel LEDs will indicate the last stored state and the **Unlocked** LED will light. The screen will then display the **Status Screen**, for example:

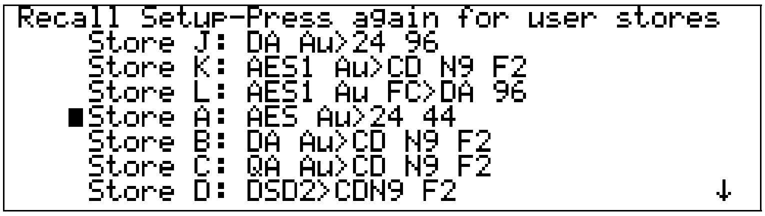


do this: Connect a signal source to an appropriate input: **AES 1, 2, 3** or **4**, **SPDIF1**, **2** or **3** or **SDIF-2 CH1 IN**, **CH2 IN** & **WCLK IN**.

We recommend an AES3 source into the **AES 1** input until you are comfortable with the unit. The procedure below assumes you have connected your source to the **AES 1** input.

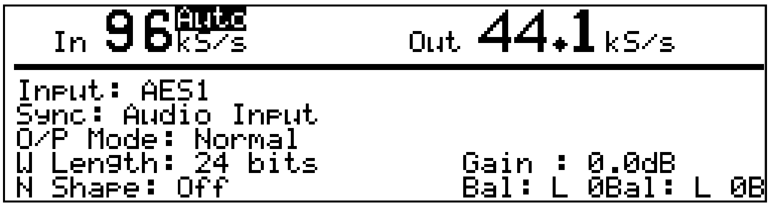
do this: Press the **Recall** button. When the display changes to the Recall screen, press the **Recall** button again.

The display will change to:



do this: Select Store A (the default setting) with the rotary control and press the **Enter** button.

A message window will appear in the display to confirm that the unit is reading the setup. Then, when complete, the message will go away, the unit will lock and the **Unlocked** LED will go out. The display then shows the status screen for the default setup as overleaf if you have a 96 kS/s source connected to **AES 1** – otherwise the left hand number will adjust to what you have connected.



The (converted) input signal will now be available at all of the outputs.

do this: Connect the output of your choice to the input of a 44.1 kS/s recorder, DAC or other equipment you wish to drive. It will now lock to the ***dCS 974***.

Note that all the outputs are active when the ***dCS 974*** is locked. They may all be connected to external equipment simultaneously if required. Similarly, all of the inputs may be connected as the active one will be selected by the Audio Input Select menu.

Now you will need to familiarise yourself with how the control software and menu system work.

do this: Read the short section on **Navigating through the Menu – what the On-Screen symbols mean,** page **36** so you know how the buttons and cursor work.

You may also find it convenient to refer to the **Quick Start Guide** while you are getting to know the unit.

Installing the Unit in a Rack

The unit is supplied with 19" rack mount ears fitted. If it is to be installed in a 19" rack, the ears supplied may be used to locate it in the rack - but:

IMPORTANT! *The ears should not be used as the only mechanical support. The unit should rest on a shelf, or be supported in some other way. The ears will just locate it in the rack, and stop it sliding forwards.*

If the unit is not to be rack mounted, the ears may be removed.

Contact *dCS* on + 44 1799 531 999
(inside the UK replace + 44 with 0)
email to: more@dcsltd.co.uk
web site: www.dcsltd.co.uk

OPTIONS

The following options may be fitted to new units or retrofitted at a later date.

Option code	Option
VID	Locking to video sample rates
V5	Mains voltage set to 230/240V
V4	Mains voltage set to 215/220V
V2	Mains voltage set to 115/120V
V1	Mains voltage set to 100V

Table 17 - Options available

Locking to Video Sample Rates

We can fit additional video frequency VCXO's (enabling frequencies such as 44.056kS/s and 47.952kS/s). This option must be fitted at ***dCS*** to allow full checking. Contact ***dCS*** for details

Mains Supply Voltage

Any unit may be set for operation from 230/240V, 220V, 115/120V or 100V A.C. The voltage setting can be updated later by your Distributor, if necessary. Specify the new voltage code.

Ordering Options for a New Unit

To order any option, check the option codes from **Table 17** above and tell us:

dCS 974 with options <option codes>

IMPORTANT! *You must specify a supply voltage option code.*

Having Your Options Changed

dCS support modifications, updates and option changes to supplied units. Major changes are normally carried out at ***dCS*** as we have extensive test facilities and can verify the changes. Please contact your Distributor or ***dCS*** for details.

IMPORTANT! *Please do not attempt the changes yourself. The unit's performance or reliability may be impaired and the warranty will be invalidated.*

MAINTENANCE AND SUPPORT

Hardware

Service & Maintenance

dCS audio products are designed not to need regular maintenance, and contain no user serviceable parts:

- there are no moving parts,
- there are no short life or wear-out parts used,
- the units have no holes through which liquids or contamination can normally enter,
- no dust deposits build up to degrade performance.

All parts are replaceable or upgradeable by *dCS*, for a period of at least five years from the original purchase date. If your unit is damaged in some way, please contact your Distributor or *dCS*.

User Changeable Parts

There are no user serviceable parts inside the case. Routine maintenance is not necessary and repairs are generally carried out by *dCS*, since this allows us to thoroughly verify the results before shipment.

There is a mains fuse in the mains socket, accessible from the outside of the unit. This may be changed by the user. The current consumption of the unit is very low (190 mA at 115 V) so it only blows if there is a fault - usually if the unit is set to its low voltage setting (100 - 120V) but has been plugged into a high voltage supply (220 - 240V). Usually no other damage is caused, but if the fuse blows repeatedly on replacement, some other damage will have been done and the unit must be returned to *dCS* for repair.

Fuse Type : 20 x 5mm 2 amp HRC fuse

If the fuse should fail, it is essential that it be replaced with one of the same type. Failure to do so could result in damage to the unit and may invalidate the warranty. To gain access to the fuse, remove the IEC mains connector, use a small flat bladed screwdriver to pry up the tab on the fuse carrier and pull it out. Push the fuse out of the clip in the carrier and replace it with a new one. Push the carrier back into the unit so that it clicks home.

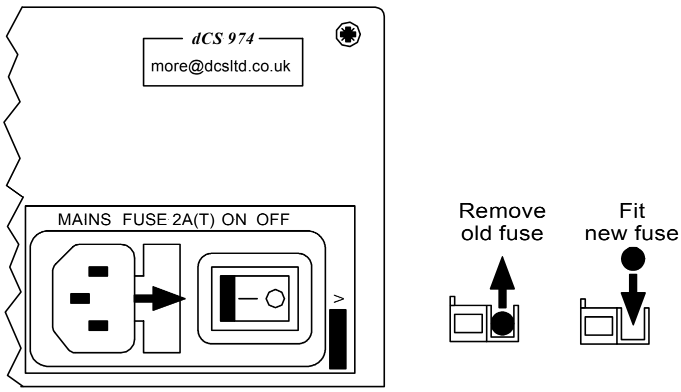


Figure 51 – Changing the Mains Fuse

IMPORTANT! *Disconnect from the supply before changing the fuse.*

Software

Installing New Software

There are two major components of software in your unit. These are the operating software, and the display software. The operating software can be downloaded via the RS-232 link from a PC COM port, using the Windows Remote software running on the PC.

To use this, follow the installation instructions on the floppy discs the remote software is supplied on. Then, run the remote programme, with whatever units you want connected. The software will scan the RS-232 chain for units (this takes a while) to see what is connected. It then reports back and for each there is an info button. This gives you the option of installing new software in that unit.

The display software comes in an EPROM, which has to be manually installed. The top cover has to be removed, the old EPROM on the front panel board removed from its socket (¼ of the way from the right hand edge, looking at the unit from the rear), and the new one installed. Make sure it is installed the right way round!

IMPORTANT! *Disconnect from the mains before removing any covers or changing the fuse.*

Warranty

Initial Warranty

dCS Ltd. warrant this product against defects in materials and workmanship for a period of 90 days from receipt by the end user. Warranty repairs must only be carried out by ***dCS*** or an authorised ***dCS*** Distributor.

Extended Warranty

The period of the Warranty cover may be extended to 1 year from receipt by the end user at no extra cost, by registering your purchase with ***dCS***. Your reseller should have arranged this on your behalf by filling in an **Owner Registration** form at the time of sale and returning it to ***dCS***. If you do not receive an **Extended Warranty Certificate** for this unit within 30 days of purchase, please note the serial number on the underside of the unit and contact ***dCS***.

Registration ensures you will receive information on important hardware and software upgrades as they become available.

If you sell the unit within the first year, the balance of the Extended Warranty may be transferred to the new owner by completing the **Owner Registration Transfer** form, page **131**, and returning it, with the **Extended Warranty Certificate**, to ***dCS***.

Warranty Exclusions

The Warranty on this ***dCS 974*** shall be void if:

- the product is misused in any way.
- any unauthorised modifications or repairs are carried out.
- the product is not used in accordance with the **Operating Conditions** stated on page **92** of this manual**.**
- the product is serviced or repaired except by an authorised ***dCS*** Distributor.
- the product is operated in a system without a mains earth (or ground) connection.
- the unit is returned inadequately packed .

Obtaining Service

If you experience problems with your ***dCS 974***, you should check the **Troubleshooting** section on page **120.** You may be able to resolve the situation yourself (for example, by changing a menu setting).

If this does not resolve the problem, contact your local authorised ***dCS*** Distributor for advice, quoting the model, the serial number from the underside of the unit, the software version number (see the **Info Submenu**, page **56**) and giving a detailed description of the fault. Please do not return any unit to ***dCS*** without obtaining a Service Return number as this will delay the repair. You may also incur costs if the unit is found to have no fault. When returning a unit, the original packing should be used to avoid transit damage. Replacement packaging sets may be purchased from ***dCS.***

During the Warranty period, there will normally be no charge for repair or replacement.

Update or Calibration

You may wish to have your unit updated occasionally. ***dCS*** offer this service - we will install any modifications or updates that have occurred since your unit was first shipped, and give the unit a full retest to current standards. The price will depend on the hardware changes necessary – so contact your dealer or us. In order to ensure speedy turn around please contact us prior to returning the unit.

Safety and Electrical Safety

There are no user serviceable parts inside the ***dCS 974*** and so there is no need to remove the covers, apart from front panel software updates. If for some reason you do:

IMPORTANT! *Disconnect from the mains before removing any covers or changing the fuse.*

There are no substances hazardous to health inside the ***dCS 974***.

TROUBLESHOOTING

FAQs

The unit fails to lock to a source signal.

- Check that Audio Input Select is set to the correct input.
- Check that Input Sample Rate is either set to Auto or set to suit the source device.
- Check that Sync Source is set correctly - try setting it to Audio Input.
- Check for damaged cables.

The unit locks but no signal is received.

- Display the Bit Activity Monitor to see if a signal is available at the unit. If there is no signal into the unit, and Detect Silence is on, there will be no signal out – even with Dither and Noise Shaping on.
- If the unit is locked to an external clock, check that Audio Input Select is set to the correct input.
- If SDIF-2 is used, check that **CH1 IN** and **CH2 IN** (upper block) are both connected.

The destination device fails to lock to the unit.

- Check that the unit is locked to a source.
- Check that Output Sample Rate and Output Mode suit the settings on the destination device.
- Ensure the destination device is set to sync to the ***dCS 974*** and not to a Master Clock.
- Ensure the destination device is capable of locking to the selected Output Sample Rate.
- Check for damaged cables.

The destination device locks but no signal or just noise is received.

- Display the Level Meters to ensure signal is present on the outputs.
- Check that the Test Generator is off by pressing the **Status** button.
- Ensure the destination device is set to sync to the ***dCS 974*** and not to a Master Clock.

The destination device connected to the AES or SPDIF outputs reports an error.

- Check that AES and SPDIF Messages are correctly set up.
- Check for damaged cables.
- Check that the destination device can receive the sample rate the ***dCS 974*** is putting out, in the appropriate (1 wire, 2 wire, 4 wire) format.

Output audio quality is poor.

- Check that Output Wordlength is set correctly.
- If truncating from 24 bits, use Noise Shaping and/or Dither.
- Ensure the unit is locked to the source device or an external clock driving the source device.
- Converting from a **very** low sampling rate to a high sampling rate cannot give sparkling hi-fi performance, although it can improve things a bit (the “upsampling” phenomenon). The ***dCS 974*** can convert a 12 kS/s, 8 bit signal up to 192 kS/s, 24 bit in real time but the linearity will be poor due to lack of source information, the noise floor will be high and the audio bandwidth will still be 5kHz (i.e. not much better than a telephone line).

Output audio bandwidth is low.

- During sample rate conversion, the signal is digitally filtered - this is essential for good performance. The output audio bandwidth will **usually** be slightly less than half the lowest sample frequency involved in the conversion.

A stereo output signal is expected but signal is present on one channel only.

- Check that Balance is not set to one side.
- If PCM SDIF-2 or DSD SDIF-2 is used, check that **CH1 OUT** and **CH2 OUT** are both connected.
- Check you are not using a Dual AES mode with just one cable connected.

A stereo output signal is expected but one channel of the input appears on both channels of the output.

- Check that the receiving unit expects two wire AES or SPDIF mode, rather than double speed, single wire.

Tone appears on the outputs.

- Press the **Status** button and check that the Gen.On label is NOT visible. If it is, select Test Mode and set the Test Generator to Off.
- Low level "idle tones" sometimes appear on the outputs if the signal on the input drops to a level below the output word length and Noise Shaping is on with no Dither. This may arise during editing sessions, for example, where digital silence is coming in to the unit, but the Detect Silence function is off. This situation will usually not arise if an unedited input from an ADC is being used, although even in these circumstances, enough noise shaping must be used. Consider using either Dither or Detect Silence or both.

The LCD display is unreadable but the Power LED and possibly some other LEDs are lit.

- Display Contrast may have been set to minimum. Switch off the unit, wait 10 seconds, hold down the **Status** button and switch on. The unit will power up using default settings, including high contrast. Alternatively, Store A is pre-loaded with default settings. Press **Recall** slowly twice, then press **Enter** to recall from it.

Strange characters appear on the LCD display.

- This can be caused by power line drop-outs or switching the unit off and on rapidly. Switch off, wait 10 seconds then switch on - normal operation should be resumed.

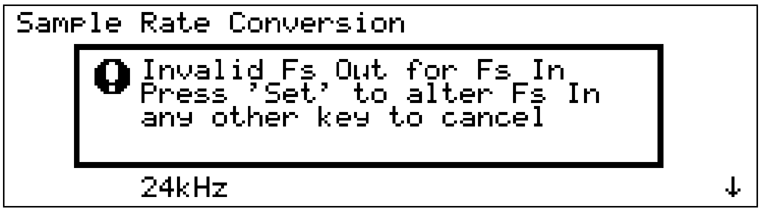
The stereo image is poor or out of position.

- Check that Phase and Swap Channels are set to Normal.
- Check that Balance is correctly set.
- It may be that the input signals are out of phase, swapped or out of balance. Use the units controls to correct this.
- If using the DSD/SDIF interface, the **CH1** & **CH2** cables may be swapped.
- If you are using a Dual AES or Quad AES mode from some other equipment, make sure you used the right outputs from that.

Maximise fails to increase the signal level to full scale.

- The range of the Maximise function is limited to +12dB. If the input signal peaks are smaller than -12dB0, Maximise will apply +12dB of gain. A second pass will apply up to 12dB more gain.

“Invalid Fs Out for Fs In” message.



- You have selected a combination that cannot be handled in one pass - see the table in the section **Output Sample Rate** on page **43** for valid single pass combinations. Choose an intermediate sample rate and convert in two passes. For example, if you need to convert from 32 kS/s to 11.025 kS/s, first convert to 44.1 kS/s and record then convert the recording from 44.1 kS/s to 11.025 kS/s.
- While changing frequencies, you have accidentally selected an intermediate combination that cannot be handled in one pass. Press **Set** and choose a valid Input or Output Sample Rate (as appropriate) from the list.

The Customise Display menu will not allow another option to be set.

- If you check the list you should find that the maximum of 5 parameters are already marked with a **x**. Remove one you do not need and try again.

If You Need More Help

Contact ***dCS***. Our office hours are 8:00 a.m. to 5:00 p.m. Monday to Friday, UK time (UTC or UTC + 1hr). Contact us by phone or fax on:

	Inside the UK	Outside the UK
Telephone	01799 531 999	+44 1799 531 999
Fax	01799 531 681	+44 1799 531 681

Table 18 – ***dCS*** Phone Numbers

You can write to us at:

dCS Ltd
Mull House
Great Chesterford Court
Great Chesterford
Saffron Walden CB10 1PF
UK

Our E-Mail address: more@dcsltd.co.uk

Our web site is: http://www.dcsltd.co.uk

Other Information

dCS produce technical notes from time to time. If you are interested in these, please do not hesitate to contact us or check our web-site.

INDEXES AND SOFTWARE VERSION NUMBERS

This manual is for software version 1.0x. v1.0x versions differ only in a few minor bug fixes.

Definitions of Units

dB0 Level in decibels, referred to a full scale sine wave in a sampled system. So, 0 dB0 is full scale.

$dB0_{DSD}$ Level in decibels, referred to the defined full scale sine wave in a DSD system. So, 0 $dB0_{DSD}$ is the zero dB audio reference level.

dBu Level in decibels, referred to a 0.775V rms sine wave, with no external loading (u = unloaded). The level of 0.775V is derived from the older dBm, for which the reference level is 1mW of signal power into a 600Ω termination from an output with 600Ω source impedance.

kS/s Sample rate in kilo-samples per second. This replaces kHz which is technically incorrect when referring to sample rates.

SQNR Signal to Q noise ratio

Tables

Figures

Keywords and Phrases

T

U

V

W

✂

2nd Owner

Owner Registration Transfer

To transfer the balance of the Extended Warranty to a new owner, please complete the form below in block capitals and forward it, with the **Extended Warranty Certificate**, to ***dCS*** at the address below within 30 days of the sale.

dCS Ltd
Mull House
Great Chesterford Court
Great Chesterford
Saffron Walden
CB10 1PF
U.K.

Serial No: 974-

(located on the underside of the unit)

Name of new owner:	
Company name:	
Address 1:	
Address 2:	
City:	**Post code:**
State:	
Country:	
Tel:	**Date purchased:**
Fax:	
E-mail:	
Website:	

Name of registered owner:

Signed:

✂