dCS 954

Digital to Analogue Converter

User Manual

Standard software version 1.5x
P3D software version 1.36

June 2000

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

PRODUCT FEATURES

Formats

- DSD, and PCM from 192 kS/s down to 32 kS/s
- Data formats supported are: AES/EBU (XLR and BNC), Dual AES (XLR), Quad AES (XLR), AES data at TTL levels, and SDIF-2 (PCM and DSD), SDIF-3 (DSD), DSD packed into 4 AES links
- P3D option: DSD packed into 3 AES links

Syncing

- Can sync to Word Clock or AES reference, or input signal, and sync to video option available

Functions

- Very high performance DAC, free from gain ranging
- High quality VCXO internal clocking
- Multichannel Sync capability
- High speed or dual AES (88.2 kS/s, 96 kS/s)
- Dual or Quad AES (176.4 kS/s and 192 kS/s)
- DDC mode converts Dual AES at 88.2 to 192kS/s or Quad AES at 176.4 or 192kS/s to High speed Single AES

Test Generator

- High quality (160 dB) signal generator with mHz resolution. Can be noise shaped truncated

Ease of Use

- Remembers last settings
- Lockouts
- Software upgrade-able without opening the box
- Can be remote controlled from PC

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About this Manual

Note that there is a fuller Contents at the end of the manual (page 82), along with an index and lists of figures and tables.

References to other sections in the text have the **"Section Name"** page … in quotation marks and bolded.

IMPORTANT!

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

The manual covers standard units and units with P3D option. P3D is a DSD data format, and these units have changed internal hardware to accommodate it. Information that is specific to P3D units is greyed.

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.

USING YOUR *dCS 954* FOR THE FIRST TIME

Product Overview

The ***dCS 954*** DAC (Digital to Analogue Converter) is a high performance converter intended for studio and live recording applications. It is designed to produce very high standard analogue output from high quality digital data formats (for example, 192 kS/s or DSD) or standard formats (for example Red Book CD or 24/96). AES3, SDIF-2 PCM formats and several DSD formats are all supported. Multiple units may be slaved to a master clock for stable multi-channel operation.

The unit is mains powered and is housed in a 1U (1.75") 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. 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 "front panel lockout" feature.

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

What's in the Box?

The contents of the box are at least:

dCS 954
User Manual
Function Menu Guide
Mains Lead
2 Spare Fuses
Remote cable
Remote software

Mains Voltages

The ***dCS 954*** is shipped with its mains 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, contact your dealer or ***dCS***.

IMPORTANT! *The **dCS 954** must be used with a mains earth!*

[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.

Installing Unit in a Rack

The unit is supplied with 19" rack mount ears fitted. If it is to be mounted in a 19" rack, the ears supplied may be used to locate it in the rack and stop the unit sliding forward – but they are not strong enough to support the unit.

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.

Getting Started

Here's what to do:

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

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

do this: If it is, using the lead supplied, connect the unit to the mains - connect no other leads at this stage - and switch on.

The 7 segment display will briefly show:

- - - - 4

and then indicates that it is out of lock:

o u t

do this: Ensure your system volume is set to a low level, then connect the analogue outputs (either balanced or unbalanced) to the inputs of your pre or power amplifier.

do this: Connect the digital output of a CD player or recorder to the AES1 input and if the AES1 input is not already selected, press the **AES1** button to select it.

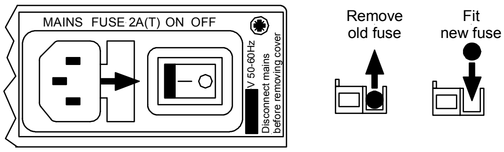

Figure 1 – Playing a CD

The unit will detect the sample rate, lock to it and unmute. The display will show the sample rate, for example:

4 4. 1

do this: If the **Mute** LED is still lit, press the **Mute** button once.

You should have audio.

Note that the balanced and unbalanced outputs are active simultaneously on the *dCS 954* and may both be connected to external equipment simultaneously if required.

Now you will need to familiarise yourself with how the front panel controls and the menu system work.

do this: Read the short section on **"The Software – the Menu"** on page 18 so you know how the buttons and menu work.

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

THE HARDWARE – CONTROLS AND CONNECTORS

Rear Panel

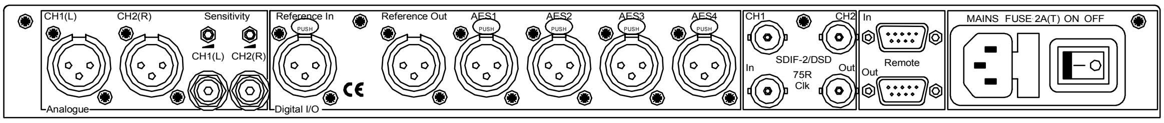

Figure 2 – Rear Panel

All input and output connectors are mounted on the rear panel. Individual connectors are clearly identified by the panel legend. Viewed from the rear from left to right, the connectors are as follows:

Balanced Analogue Outputs **3 pin XLR male (2 off)**

Unbalanced Analogue Outputs **RCA phono (2 off)**

Output Level Adjustment **(trimmers)**

Two multi-turn potentiometers set the full scale output levels for the Balanced Outputs only. These are factory preset for full scale with output levels of +14dBu. If necessary, adjust with a suitable trim tool or a small flat-bladed screwdriver. Turn clockwise for increased gain. Take care to ensure the stereo outputs remain in balance. The trim range is ±6dB.

Reference In **3 pin XLR female**

Reference Out **3 pin XLR male**

Reference In is an AES/EBU reference input for synchronising the unit to a Master Clock. Reference Out is an unbuffered loop through, directly coupled to it, for use in a reference daisy chain. A terminating resistor may be turned on or off, using the menu (see **Ref In** command, page 23), if several units are to be daisy chained with the same word clock.

AES1, 2, 3 & 4 Digital Inputs **3 pin XLR female (4 off)**

Four AES/EBU inputs which may be used independently or in groups of two (Dual AES on AES1 & 2 or on AES3 & 4) or four (Quad AES or 4-wire DSD).

P3D units will also accept DSD in P3D format connected to AES1, 2 & 3.

SDIF/DSD CH1, CH2 Data **BNC (2 off)**

These BNC connectors can be both inputs and outputs. In normal operation they are inputs for SDIF-2 encoded PCM, or SDIF-2 or SDIF-3 encoded DSD. They are both TTL level signals for a 75 ohm line. They can be set to accept TTL level AES3 coded signals, using the menu (see BNC I menu command, page 26).

In addition, they can be used as data outputs, for re-formatting DSD data, in DDC mode. See the Ref In menu command on page 23.

SDIF/DSD Clk In **BNC**

SDIF/DSD Clk Out **BNC**

This pair of BNC connectors normally take in and give out Word Clock. The functions are set by the menu. Clk In is terminated and Clk Out is regenerated internally, so these lines can be used for daisy chaining many units together, without loading problems. See Figure and Figure for the time alignment of these signals.

Additionally, they can both be set for TTL level AES3 coded signals, using the menu. The input connector is controlled by the BNC I menu command – see page 26 – and the output connector is controlled by the BNC O menu command – see page 27. As an AES output, it will output the signal on the currently selected input (whatever is playing). The input may be just a clock, for locking purposes, or a full AES3 coded input. The choice is controlled by the BNC menu command – see page 27 – and the input is selected by the **BNC** button (page 12).

Remote In & Out **9 pin D type male (2 off)**

If the Windows™ Remote software is in use, connecting Remote In to a com port (RS-232 port) on a PC running the Remote Control program allows the unit to be controlled by the PC. Remote Out may be connected to another suitably equipped *dCS* unit, allowing several units to controlled by the same PC with one RS-232 daisy chain. In addition, the unit may be software upgraded without removing the lid by downloading new software via the Remote In port – see **"Installing New Software"** on page 73.

Connect up Remote ports using a 9-way screened cable, fitted with 9-way 'D'-type connectors at each end, wired pin 1 to pin 1, pin 2 to pin 2, etc. The same type of cable can be used unit to units as com port to first unit. Suitable cables are available from *dCS*.

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

Switched, fused and filtered IEC mains connector.

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 954*
Manufacturers Name and Country of origin (*dCS* Ltd, UK)

Serial Number

The underside of the unit will have a label on that contains a number such as 954 4B1 6B2 2A1 3A2 12345. This is the 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 you software updates.

Front Panel

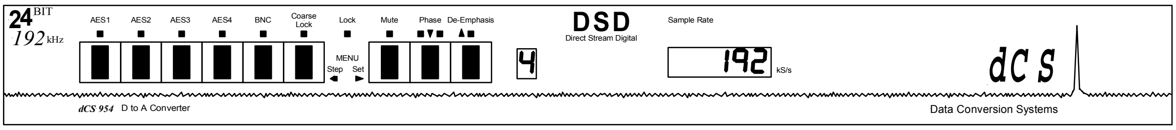

Figure 3 – Front Panel

The ***dCS 954*** uses a combination of front panel buttons for frequently changed functions and a step through menu for features you might set and forget.

AES1, AES2, AES3, AES4 & BNC **Menu Step**

The 5 buttons on the left side of the front panel select the active input(s). The LEDs above these switches indicate the input status as follows:

LED state	Function
Bright	Source available and selected. (More than one for multi-wire.)
Dim	Source available but not selected.
Off	Source not available.
Flashing	Source selected but not available.

do this: Connect the source equipment to the unit as necessary. The unit will indicate active inputs by dimly lighting the appropriate LED.

do this: To manually select a single input, press the appropriate button, hold it down for a second then release.

The LED over the button will brighten and the unit will attempt to lock to that input. The **BNC** button selects the SDIF input. Note that this can be either SDIF (2 or 3 if DSD is being used, automatically sensed) or TTL level AES3, under menu control. See menu commands BNC, BNC I, BNC O on page 26 onwards

do this: To select a Dual AES input on AES1 & 2, press the **AES1** and **AES2** buttons together, hold for a second and release the **AES2** button first. Similarly for Dual AES on AES3 & 4, press the **AES3** and **AES4** buttons together, releasing the **AES4** button first.

The two LEDs over the chosen buttons will brighten and the unit will attempt to lock to the Dual AES input.

do this: To select a Quad AES input on AES1, 2, 3 & 4, press the **AES1** & **AES4** buttons together, hold for a second and release the **AES4** button first.

The four LEDs will brighten and the unit will attempt to lock to the Quad AES input.

IMPORTANT! *Take extra care when connecting Quad AES as it is very easy to connect the wires in the wrong order. Although dCS puts messaging in data streams to allow equipment to sort this sort of problem out itself, not all other manufacturers do. If there is no messaging in the data stream, the only indication that this has happened is poor audio quality. Labelling the cables is a sensible precaution.*

IMPORTANT! *If the selected format does not match the source(s) connected, the audio output may be severely aliased mono or aliased mono mixes of the sources. This should pose no risk to ears or speakers (assuming the system gain is set sensibly) but cannot be detected by the unit without correct messaging.*

The unit stores the last input selection at power down and re-loads it when power is restored. For example, if the previous setting was Quad AES, the unit will return to this mode at power up, provided all 4 AES inputs are valid and carry the same sample rate. The unit will detect the sample rate present on AES1 and lock to it. If one or more inputs are disconnected, the unit will lock to Single AES on AES1 until the required inputs are available.

Your ***dCS 954*** can automatically select the data format. For details, see the menu option I For on page 25.

To use the unit in DSD mode, see the menu entry for DSD on page 20.

The **Active Input** button is the source selector button below a bright LED or, if the unit is unlocked, a flashing LED. On its own or with the other source selector buttons, the **blue** text on the panel applies. With the other menu buttons (**white** type on the front panel) it is the menu **Step** button.

For Menu operation as the **Step** button, see the section "**The Software – the Menu"** on page 18.

Coarse Lock

Some digital audio equipment (even some quite expensive products) produces data streams with a level of jitter outside the AES3 specification. In particular, sources that involve mechanical movement between tracks (for example, some CD players) can show large timing transients as the movement occurs, which can upset the ***dCS 954***, causing intermittent data errors or muting. To cater for this situation, the dCS 954 allows two PLL settings – fine lock and coarse lock. If the problem arises, pressing the **Coarse Lock** button forces the unit to use the coarse lock mode, which may solve the problem. The LED indicates that coarse lock is selected.

Unless necessary to maintain a stable lock, coarse lock should be turned off it will cause degraded audio quality. It should only be used where degraded audio is better than no audio!. For more information on this topic, see section **"Jitter and PLL bandwidths"** on page 68

Note that the use of unscreened digital audio cables can occasionally cause the unit to mute and re-lock, if they allow in gross interference from some nearby source.

Lock Indicator

When lit, this indicates that the front panel controls are locked out to prevent accidental or unauthorised operation. Lock is turned on and off using the Loc menu option, see page 28.

Mute **Menu Set**

The **Mute** button is dual function – on its own (**blue** type on the front panel), it mutes and unmutes the analogue outputs when the unit is locked to a source. With the other menu buttons (**white** type on the front panel) it is the menu **Set** button.

The analogue outputs are automatically muted at power up and remain so until lock is achieved, as indicated by the LED above the **Mute** switch. If **"A-Cut"**, page 22, is turned on, the unit will mute automatically if the source data is identified as non-audio.

For Menu operation as the **Set** button, see **"The Software – the Menu"** on page 18.

Phase **Menu Down**

The **Phase** button is triple function – on its own (**blue** type on the front panel), it reverses the phase of the analogue outputs. If the "**Panel**" option, page 27, is set to **Volume,** it functions as the Volume down button. With the other menu buttons (**white** type on the front panel) it is the menu **Down** button.

The left hand LED above the **Phase** button lights if **CH1(L)** outputs are inverted, the right hand LED lights if **CH2(R)** outputs are inverted. Pressing the **Phase** button toggles both channels between absolute and inverted phase. Holding the button down causes the unit to cycle through the following sequence:

LEDs		Ch1 (L) phase	Ch2 (R) phase
on	off	inverted	absolute
off	on	absolute	inverted
off	off	absolute	absolute
on	on	inverted	inverted

Table 1 – Phase LEDs and Channel Phasing

Release the button when the required setting appears.

For Menu operation as the **Down** button, see **"The Software – the Menu"** on page 18.

De-Emphasis **Menu Up**

The **De-Emphasis** button is also triple function – on its own (**blue** type on the front panel), it sets the De-Emphasis characteristic. If the "**Panel**" option, page 27, is set to **Volume,** it functions as the Volume up button. With the other menu buttons (**white** type on the front panel) it is the menu **Up** button.

De-Emphasis is available for 32, 44.1 and 48kHz sample rates only. Pressing the button repeatedly causes the single digit **Mode** display to the right of the button to cycle through the following options:

Display	De-Emphasis in Use (low sample rates)
A	Automatic - the unit automatically implements the De-Emphasis characteristic coded in the data stream. The display changes to 5 or C when De-Emphasis is automatically applied.
5	The unit implements 50/15μs De-Emphasis.
C	The unit implements CCITT J17 De-Emphasis.
-	De-Emphasis disabled.

Table 2 – Emphasis Indication, low sample rates

dCS recommend setting the unit to **A**utomatic unless there is an error in the De-Emphasis message in the incoming data stream.

For Menu operation as the **Up** button, see the section "**The Software – the Menu"** on page 18.

Mode Display

At sample rates over 48kS/s, the single digit LED **Mode** display to the right of the **De-Emphasis** button shows the input format:

Display	Input Format
1	Single AES
2	Dual AES
3	P3D mode
4	Quad AES or DSD 4-wire

Table 3 – Output Data format indication, higher sample rates

Sample Rate Display

The main display generally shows the incoming sample rate, in kS/s, or the mode (DSD). When other parameters are set, it briefly shows the new setting (volume, tone frequency, etc) then reverts to its normal display. In the case of an error condition, it will display an error message.

If the unit is being slaved to a reference source, the display also indicates which reference input it is slaved to.

xxx	The sample rate, in kS/s (32, 44.1, 48, 88.2, 86, 176.4, or 192).
b xxx	Slaved to the BNC Word Clock In.
r xxx	Slaved to AES Reference In.
d xxx	Temporary display during locking – the unit has detected the base reference sample rate and is attempting to lock to it.
. xxx	Temporary display during locking – the unit is lining up word clock out to word clock in.

Important error messages are given below – a full list is given in the section **"Error Codes and Messages"** on page 76.

BadFs	The clock source is not in pull in range, or is poorly formatted. The unit cannot lock to it.
Err.xy	An error has been detected. Please refer to **"Internal Device Error Codes"** on page 76 for more specific details on error codes.
Hot	The unit is overheating, probably due to inadequate ventilation. Please check positioning and cooling.
Ouch	The "Hot" warning has been ignored and the unit is getting so hot that damage may follow.
(blank)	If the display is completely blank for any significant period, try switching off for 10 seconds then switching on again. If this does not solve the problem, contact your distributor or ***dCS***.

The display is also used for **Menu** options.

THE SOFTWARE – THE MENU

Overview

The ***dCS 954*** has many other functions that either need to be accessed only occasionally, or are informative in nature. These functions can be accessed either by the Remote software, running on a PC and connected to the unit by an RS-232 link - or (in most cases) by the **Menu,** via the front panel. If a function is set by the menu or the Remote, the unit remembers it, and it will be set this way for ever (or until you set it to something different). You can customise your unit in this way. Information-only items are displayed for a time, then the display reverts to normal.

Menu buttons are indicated by white text on the front panel. There are four:

Step	otherwise the **Active Input**. This is the input selector button with the bright LED or the one furthest to the left if more than one is bright.
Set	otherwise **Mute**
Down	otherwise **Phase**
Up	otherwise **De-Emphasis**

Entering the Menu

The Menu is entered by holding down the **Step** and then pressing the **Set** button once. The display will show:

Func

You are now in the menu, and the menu buttons now have their alternate meanings.

Moving through the Menu

Press the **Step** button again to step through the Menu items listed below. When you reach the required item, press the **Set** button to view or change its setting. This either displays the current state, or changes to the next state, or causes an information function to read out, or enters a lower level (as in the Tone generator, for example). If you have entered a lower level, pressing **Step** steps through its options. When you reach the one you want, press **Set** and then use the **Up** or **Down** buttons to increase or decrease a value (such as **Level** or **Frequency** on the Tone Generator).

If no changes are made in 5 seconds, the unit exits the Menu. When one item has been set, press the **Step** button again if you wish to continue cycling through the Menu.

There is a knack in doing this easily – once it has been gained, it becomes very easy to use the functions it accesses.

The Menu Sequence

To access the Function Menu, hold down the **Menu Step** button and press the **Menu Set** button.
(The **Menu Step** button corresponds to the selected input - LED bright or flashing.)
To step through the Menu items, press the **Menu Step** button repeatedly.
To select an item or one of its options, press the **Menu Set** button.
Use **Menu Up** and **Menu Down** buttons to alter **RS232** address, Tone **Level** and Tone **Frequency**.
To exit the Function Menu, either select the **End** item or wait five seconds.

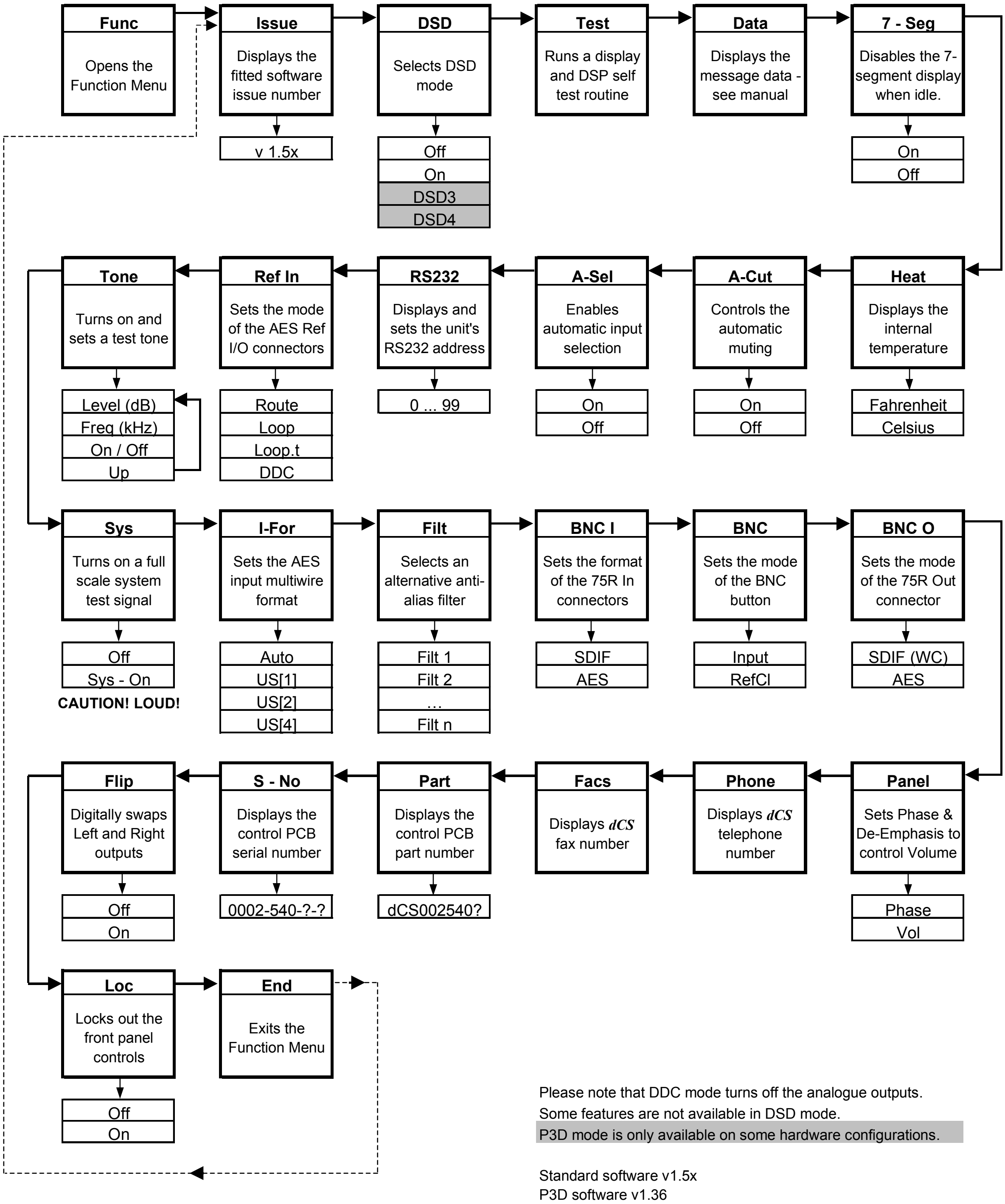

Figure 4 – Menu Sequence

Menu Items

Issue

Displays the software issue when **Set** is pressed.

DSD

Selects DSD mode. When on and locked, the unit displays "dSd". This mode takes about 15 seconds to load, during which time the menu cannot be used. There are two options:

Off	The unit operates in PCM mode.
On	The unit will accept either DSD via the SDIF input connectors (automatically selecting SDIF-2 or SDIF-3 format) or DSD packed into 4 AES3 44.1kS/s links.

P3D units have a slightly more explicit menu sequence for this item:

Off	The unit operates in PCM mode.
P3D	The unit will accept DSD via the SDIF input connectors (automatically selecting SDIF-2 or SDIF-3 format) or DSD packed in P3D format on connectors AES1-3
DSD4	The unit will accept DSD via the SDIF input connectors (automatically selecting SDIF-2 or SDIF-3 format) or DSD packed into 4 AES3 44.1kS/s links.

To select DSD on the SDIF inputs, first set the dSd menu option to On. Connect the 2 data lines plus a 44.1kS/s Word Clock and press the **BNC** button. The LED above the button will brighten and the display will show dSd.

To select 4-wire DSD mode, first set the dSd menu option to On. Connect the four-wire encoded DSD 44.1kS/s data streams to the AES1 – 4 inputs, taking care to ensure they are in the correct order, and press the **AES1** button. The 4 LEDs above the buttons will brighten, the unit will display dSd and unmute. The unit detects swapped or missing wires by displaying (for example) 1324 or 1-34.

DSD is so different from PCM that many of the PCM related features are no longer appropriate or are not implemented.

- the **Mute** and **Coarse Lock** buttons operate as usual
- the **Phase** and **De-Emphasis** buttons do not work and
- the **Volume** control is disabled.
- the following Menu items are disabled or have their functions locked: Data, A-Cut, A-Sel, Tone, Sys, I-For, BNC I, BNC, BNC O, Panel and Flip.

Filt works – there are 4 filter options in DSD mode. See the section on **"DSD"** starting on page 40 for more details – they trade-off ultrasonic signal bandwidth and out-of-band noise. A-Cut is set to On, as DSD data in multi-wire AES formats should be flagged as Non-Audio. The Ref In option is disabled in SDIF DSD mode.

IMPORTANT! *If the Non Audio flag is stripped by the recorder and the **dCS 954** is not set to DSD mode, it will accept DSD data as AES3 PCM and will output potentially damaging full scale noise.*

If you connect DSD to the SDIF input and select it while DSD mode is Off, the unit will remain muted, identify the format, automatically select DSD mode and unmute.

If you connect SDIF while DSD mode is On, the unit will remain muted until you set DSD mode to Off.

To use P3D mode (assuming your hardware is suitable), first set the DSD menu option to dSd3. Connect the three-wire encoded DSD 44.1kS/s data streams to the AES1-3 inputs taking care to ensure they are in the correct order, and press the **AES1** button. The 3 LEDs above the buttons will brighten and the display will show dSd. Contact ***dCS*** for further details.

Test

Runs a display self test routine. When successfully completed, the unit displays Pass and returns to normal operation. Otherwise an error message Err.xy is displayed – please refer to **"Error Codes and Messages"** on page 76 for more specific information.

Data

Reads and displays the message information from an incoming AES/EBU data stream. This provides a simple access to the digital audio message bits for reference or to assist with system debugging. There are separate data sets for Professional and Consumer streams - the unit reads the incoming stream and decides which it is. Pressing **Set** again steps through the message fields (see the Appendix for a fuller listing of what these are):

First field

Pro	Professional use channel status block, or ...
Con	Consumer use channel status block.

Second field

Audio	Audio data, or ...
n.Aud	Non-audio data.

Third field

N Ind	Emphasis not indicated, or...
None	No emphasis, or ...
50 15	50/15µs emphasis, or ...
J. 17	CCITT J.17 emphasis, or ...
D ----	Unable to decode emphasis, or ...
CPY P	Copy prohibit, or ...
CPY E	Copy enable.

Fourth field

Exxxx	Encoded sample rate is "xxxx", i.e. 32, 44.1, 48, 88.2, 96, 176.4 0r 192kS/s. Non-***dCS*** equipment may only decode the first 3 options. or ...
E----	Unknown encoded sample rate.

Fifth field

O.xxxx	Channel origin (alphanumeric), or ...
NonE	Emphasis not indicated, or...
50 15	50/15µs emphasis, or ...
D ----	Unable to decode emphasis.
Sixth field	
D.xxxx	Channel destination (alphanumeric), or ...
Cd	CD source, or ...
Enc	2-channel encoder / decoder, or ...
dAt	DAT machine, or ...
S----	Unknown source.

7-Seg

Disables the main 7 segment LED display. When set to Off, the Sample Rate display turns off 5 seconds after the last button press. A dot in the lower right hand corner of the display remains lit to indicate that the display has been deliberately blanked. The display springs back into life (temporarily) if the menu is used subsequently. Error or warning messages are displayed regardless of this setting. The Mode display is not affected.

Heat

Displays the internal temperature of the unit. Press **Set** to toggle between Fahrenheit and Celsius.

A-Cut

Automatically mutes the analogue outputs if the selected AES/EBU data stream is flagged as Non Audio. This gives useful protection and should normally be set to On. Set to Off to disable Auto-Muting. To warn you of this, the Mute LED will flash when the unit is not muted.

IMPORTANT!

Auto-Muting should not be disabled unless absolutely necessary as this will prevent the unit muting if data errors occur or a non-audio CD is played. Monitoring such data can cause loudspeaker and hearing damage!

A-Sel

When set to On, automatic input selection is active. The unit will detect the inputs connected to an active source and select one in the priority order AES1, AES2, AES3, AES4, BNC. If the selected input is disconnected or turned off, the unit will select the active input with the highest priority. You can select a different input by pressing the appropriate button.

You might wish to use this function when a main signal is being fed into one input, with a backup signal from another source being fed into a different input. If the main signal fails, the unit will automatically switch to the backup. Or, if you wish the dCS 954 to just play whatever you plug into it, and not worry about which input you are using, you might wish to set A-Sel to On.

If the I-For menu item (“Input Format”, see page 25) is set to Auto, the unit will read the multi-wire format from the message flags in the AES3 data streams, selecting Dual or Quad AES groups as necessary. If the message flags are incorrectly set, the unit may set the wrong multi-wire format.

If the input format is set to Single, Dual or Quad AES format, the unit will ignore the message flags and group the AES inputs accordingly.

Set A-Sel to Off to disable automatic input selection.

RS232

Displays - and allows access to – the unit's RS-232 identity code (an address between 0 and 99). This is used by the remote control software to send specific messages to specific units. Use **Up** and **Down** to change this address if you are operating several units in a multichannel set up. The RS-232 control formats and procedures are covered in more detail in section **"RS-232 Remote Control Interface"** starting on page 56

IMPORTANT! *When using the Remote Control, each unit in the daisy chain MUST be set to a different RS-232 address.*

Ref In

Displays and sets the mode of the **AES Reference In/Out** connectors. The options are:

Route	**Reference Out** (connected in parallel with **Reference In** – beware!) is internally driven with the selected AES input signal. If the input is Dual or Quad AES, the data on the lowest of **AES1** or **AES3** appears on **Reference Out**. If **BNC** is selected, there is no output.
Loop	The unit attempts to lock to **Reference In,** which is looped through to the **Reference Out**, with no termination resistor (termination is then about 1kΩ, so several units can be daisy chained).
Loop.t	As above, but terminates the input to achieve 110Ω. Use at the end of a daisy chain.
ddC	Converts the data on the selected input to single AES and sends it to **Reference Out**. The data on a Single AES input is copied with no conversion. The data on a Dual or Quad AES input is converted to Double speed Single AES. 176.4kS/s is converted to 88.2kS/s and 192kS/s is converted to 96kS/s. If **BNC** in **SDIF** mode is selected, the output is AES clock with no data at the Word Clock rate. If the input is 4-wire DSD, the data is converted to SDIF format DSD **and sent to the SDIF inputs**. **SDIF Ch1** is checked first before the SDIF inputs are changed to DSD outputs.

IMPORTANT! *In DDC mode, the analogue outputs are generally muted because the units resources are used to perform a digital to digital conversion. They are not muted if the input is 4-wire DSD.*

Tone

(**Tone** Generator). This controls an internal Tone Generator, whose level and frequency can be adjusted. Pressing **Set** enters a submenu, which accesses the following functions:

Level	The output level in dB0. It can be changed in 0.1dB steps using the **Up** and **Down** buttons. Press **Set** to accept the change. The rate of change accelerates if the button is held down. The range is from 0dB0 to –120dB0.
Freq	The output frequency, in kHz. It can be changed by using the **Up** and **Down** buttons. Press **Set** to accept the change. The rate of change accelerates if the button is held down. The step size becomes progressively smaller below 1kHz.
On/Off	Toggles whether the Generator is on or off.
Up	Allows the menu to be re-entered to set other functions. Alternatively, if left, the menu will just time out, keeping the last settings.

The frequency range is from 1Hz to 99kHz but the level rolls off at the extremes, due to anti-image filters in the analogue circuitry. The Generator output level is within –0.1dB from less than 10Hz up to 50kHz.

The Tone Generator works standalone, without a digital input. If locked, the unit will be unlocked when the Generator is turned on. At power up, the Generator is always set to Off, 1kHz and –18dB0 – this is a safety measure.

If very fine frequency resolution is wanted, RS-232 control will give mHz frequency resolution. See section **"RS-232 Remote Control Interface"** starting on page 56.

The tone generator is very pure – it is a good analogue source.

IMPORTANT!

To avoid damage to your ears, loudspeakers and power amplifiers, use the Tone Generator with care .

Sys

System phase check. This turns on a full-scale test signal for checking absolute output phase on an oscilloscope.

IMPORTANT!

To avoid damage to your ears, loudspeakers and power amplifiers, use the Sys *option with care . Turn your system volume level well down before using this feature!*

Turn this option On, the display will flash Sys On and the waveforms shown below will appear on the analogue outputs. The top waveform is from the left channel, the bottom is from the right.

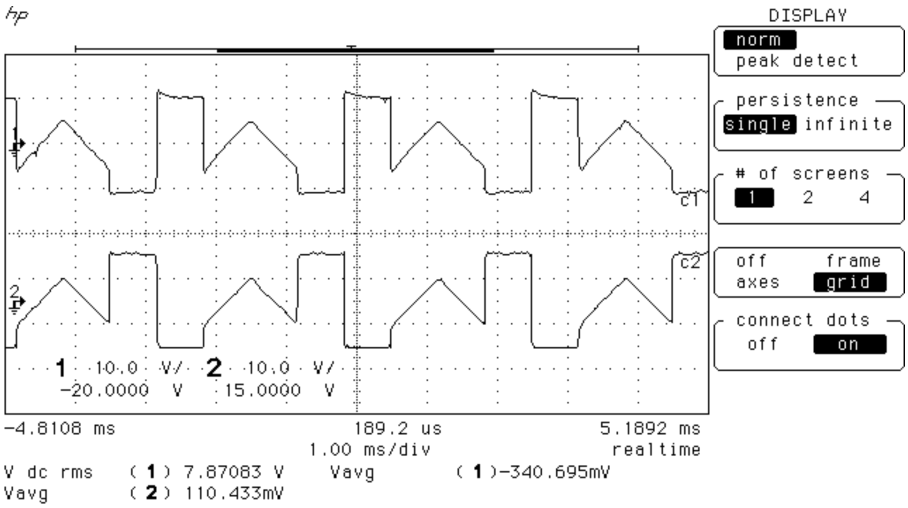

Figure 5 – In-phase Sys waveform

Referring to Figure 5, if the triangular sections point up, that channel is in phase. A triangular section pointing up with a rectangular block on its left side indicates left channel. A rectangular block on the right indicates right channel.

Figure 6 shows the waveforms with both channels inverted.

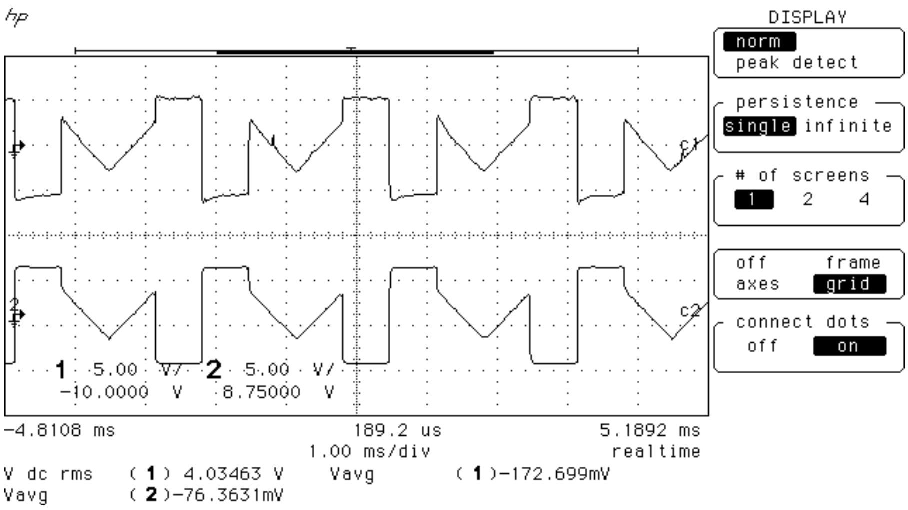

Figure 6 – Out-of-phase Sys waveform

These waveforms are not affected by the setting of the **Phase** button.

Set Sys to Off when you have finished checking. Sys is turned off at power down.

I-For

(**I**nput **For**mat). Sets the AES input multi-wire format **I**nput **For**mat. There are 4 options:

Auto	The message flags on the selected input are checked and the format is automatically set to Single, Dual or Quad AES. Dual AES must be connected to **AES 1** & **2** or **AES 3** & **4**.
US[1]	User-selected Single AES mode.
US[2]	User-selected Dual AES mode.
US[4]	User selected Quad AES mode.

You can over-ride any setting using the **AES1** – **AES4** buttons.

In Dual or Quad AES modes, the unit will warn you of missing wires with a display like 1-34 (**AES2** not connected for Quad AES). Swapped wires causes a display like 1243 (**AES3** & **4** swapped). The checking relies on correct messaging and may not work with non-***dCS*** equipment.

Filt

(**Filt**er). Selects one of several anti-image[3] filter responses. The filters should be evaluated by ear. Filt1 gives the sharpest cut off, just below half the sampling frequency. This is the normal setting. Filt2, Filt3, Filt4 give progressively more relaxed responses, slightly degrading the Nyquist image performance but sharpening the impulse response. This affects the stereo or multi-channel image. Different filters may be appropriate for different material.

A typical audio signal contains very little signal energy above 10kHz, so there is some justification for relaxing the filter attenuation as 20kHz is approached.

DSD mode also has four filters. Metering DSD is difficult because the high level of ultrasonic noise can cause spurious meter readings. DSD Filt4 is specially designed for metering the 0 – 20kHz band and may not give the best sonic performance.

BNC I

(**BNC I**nputs). In PCM mode, this sets the format of the **BNC In** connectors.

SDIF	Configures Ch1, Ch2 and Clk In as an SDIF-2 interface.
AES	Configures Clk In to accept AES3 encoded data at TTL levels at up to 96kS/s.

Select these inputs using the **BNC** button.

If the unit is already locked to SDIF or DSD, the AES option is disabled.

[3] Nyquist images, not stereo images. These are reflections of the pass-band spectrum about Fs/2 caused by any digital-to-analogue conversion process. They must be filtered out by the converter.

BNC

(**BNC** Button). In PCM mode, this controls the operation of the **BNC** button.

Input	Configures the BNC input connectors as a data input. Press the **BNC** button to select it.
RefCl	Configures the BNC word clock in (Clk In) connector as a reference clock input. Press the **BNC** button to sync the unit to the Word clock while taking data from one or more of the AES inputs. The BNC LED will light, in addition to the LED(s) for the data source. Press the **BNC** button again to sync to the data input instead.

This mode of operation enable the unit to sync to one clock source and rapidly switch between other synchronous sources without going through a PLL locking sequence. It might be used, for example, in A/B comparisons or in locking to a house sync, just taking data from AES feeds.

If the unit is already locked to SDIF or DSD, the RefCl option is disabled.

BNC O

(**BNC O**utput). In PCM mode, this sets the format of the **BNC Out** connector (Clk Out).

SDIF	Sends out SDIF Word Clock on the Clk Out connector. The unit must be locked to generate an output.
AES	Sends out AES3 coded data at TTL levels, at up to 96 kS/s on the **Clk Out** connector. This works when the unit is locked to Single AES, SDIF or BNC - AES input.

Panel

(**Panel Control**). Sets the operation of the **Phase** and **De-Emphasis** buttons.

Phase	Sets the two buttons to their standard functions.
Vol	The two buttons act as a digital volume control. **Phase** reduces the **Volume** in 0.1dB steps, while **De-Emphasis** increases the **Volume**. The change accelerates if a button is held down. The range is 0dB to –20dB.

Phone

dCS telephone number scrolls across the display.

Facs

dCS fax number scrolls across the display

Part

The control board part number (version) scrolls along the display.

S-No

The control board serial number scrolls along the display. You will need something to write this on, if you call us for help.

Flip

(**Flip** channels). Normally set to Off. If you find the Left and Right outputs from your system are reversed due to a connection error, set Flip to On to digitally swap them back. The main display shows Flip.d while this feature is turned on. Flip is not stored at power down so you should correct the error and turn Flip Off.

Loc

(**Loc**k/Unlock front panel). Front panel Lock, normally Off. Set to On to prevent unauthorised changes using the front panel buttons. The menu has to be accessed to turn the lock off again.

End

Exits the menu.

Typical Applications

Using a *dCS 954* for DSD

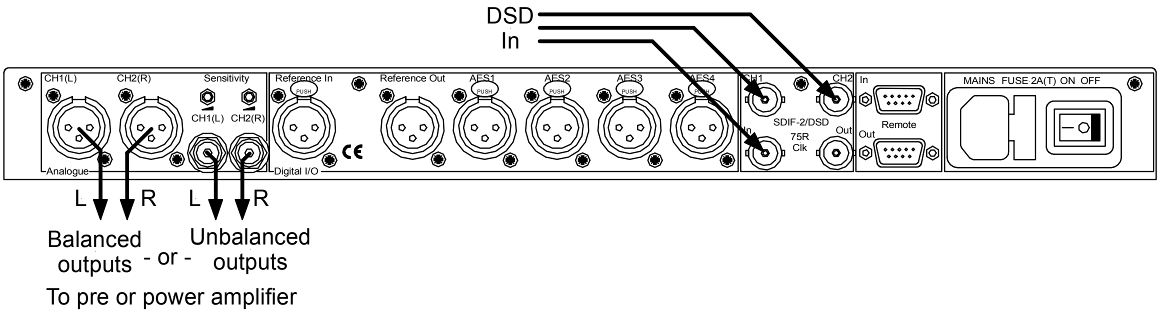

Figure 7 – DSD input configuration

do this:	Set dSd in the menu to On, press the **BNC** button.
do this:	Select a filter.
do this:	Ensure **Mute** is Off.

Using a Master Clock to Sync a *dCS 954*

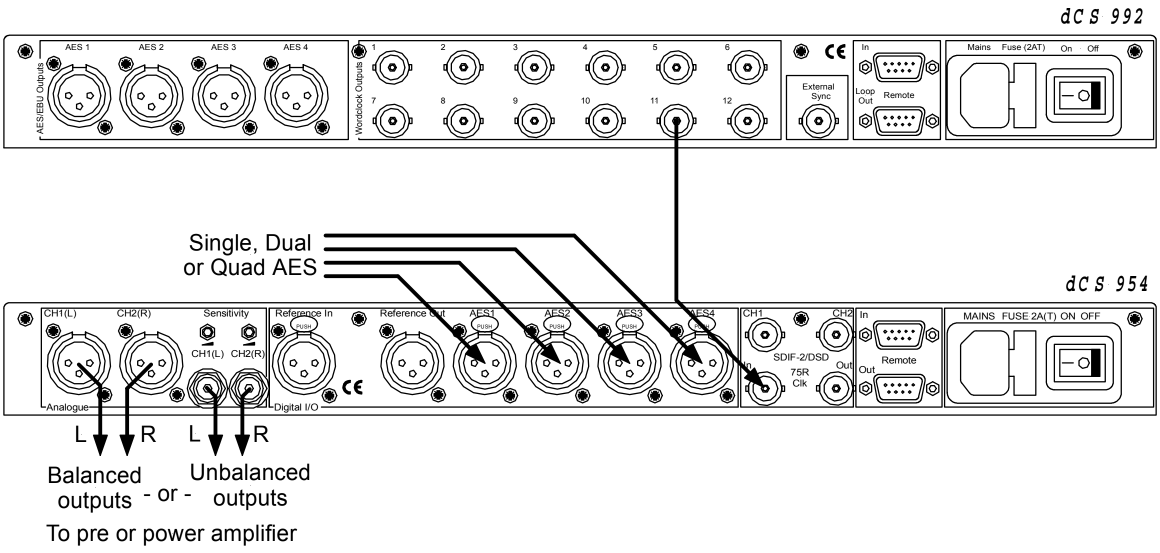

Figure 8 – Syncing a *dCS 954* to a Master Clock

do this:	Set BNC in the menu to RefCl.
do this:	Select the required input(s), then press the **BNC** button as well.
do this:	Sync the source to the Master Clock.

Replaying DSD from an 8 track 16/44.1 PCM Recorder

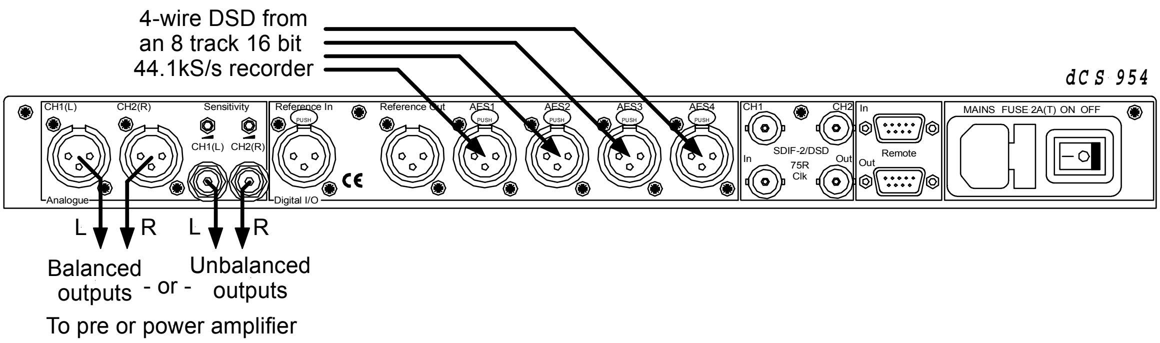

Figure 9 – Replaying 2 channel DSD from an 8 track 16/44.1 PCM recorder

do this: Make sure that the unit is in DSD mode.

do this: Connect the 4-wire DSD source, taking care to ensure the wires are in the right order.

do this: Press **AES1** to select the 4-wire DSD input, select a filter and ensure **Mute** is off.

Operating Several Units on One Remote Chain

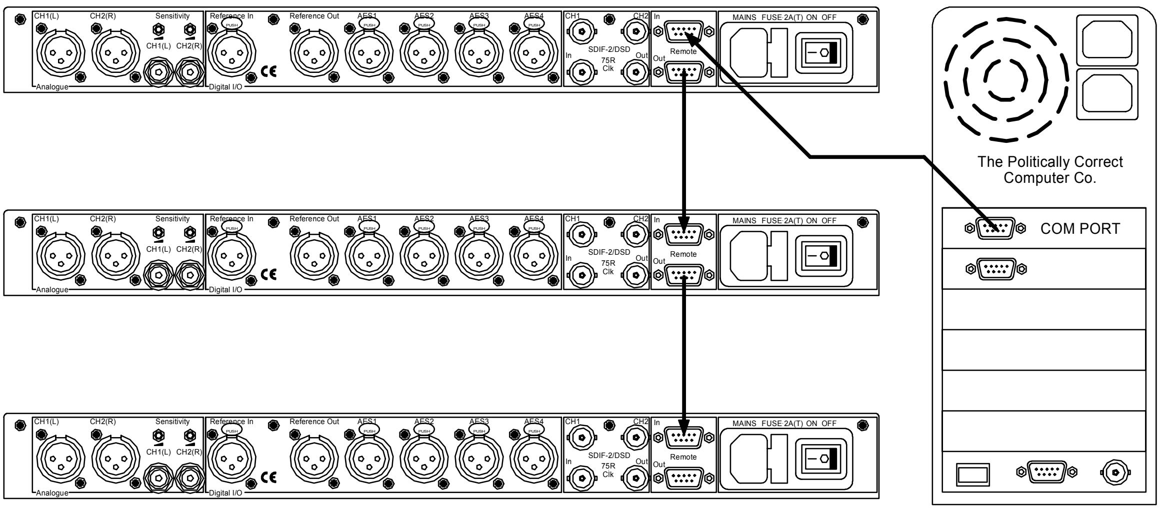

Figure 10 – Multi-unit Remote Daisy Chain

The PC can control several units (up to about 10) on each daisy chain. To make them individually addressable, each unit needs its RS-232 address to be different. They can then be identified, and grouped, in the remote window. A mixture of *dCS* unit types may be used.

See **“Remote In & Out“** on page 11 for cable details, and **“RS-232 Remote Control Interface”** from page 56 for more details.

Six Channel PCM Set Up

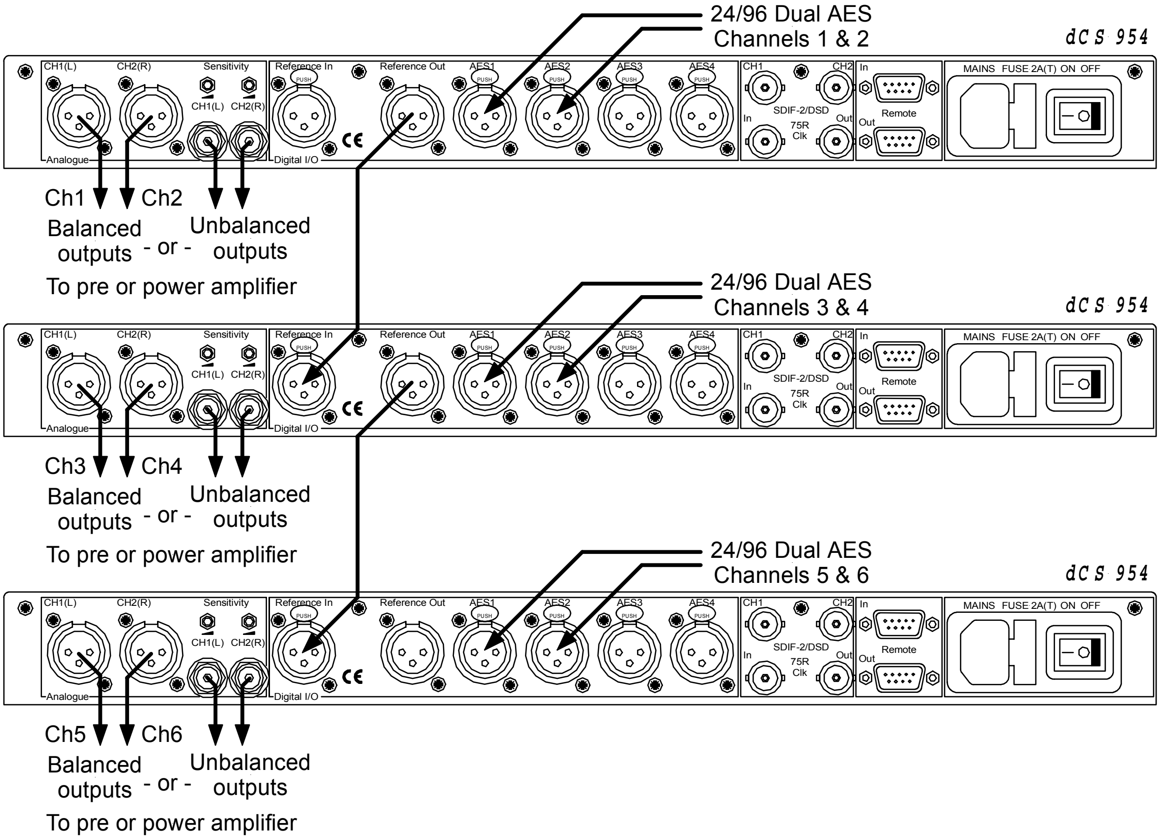

Figure 11 – Six channel set up

do this: The top ***dCS 954*** needs to have its Ref In option set to Route, the middle one to Loop and the bottom one to Loop.t.

The units self align quite accurately (see section "**Sample Alignment**" on page 41 onwards). Alternatively, Word Clock may be used as the syncing method, with no special set ups.

If the six channels are not bit-aligned, all three units should be slaved to their AES inputs, rather than **Reference In**.

Replaying 6 channel DSD from a 24 track 16/44.1 PCM Recorder

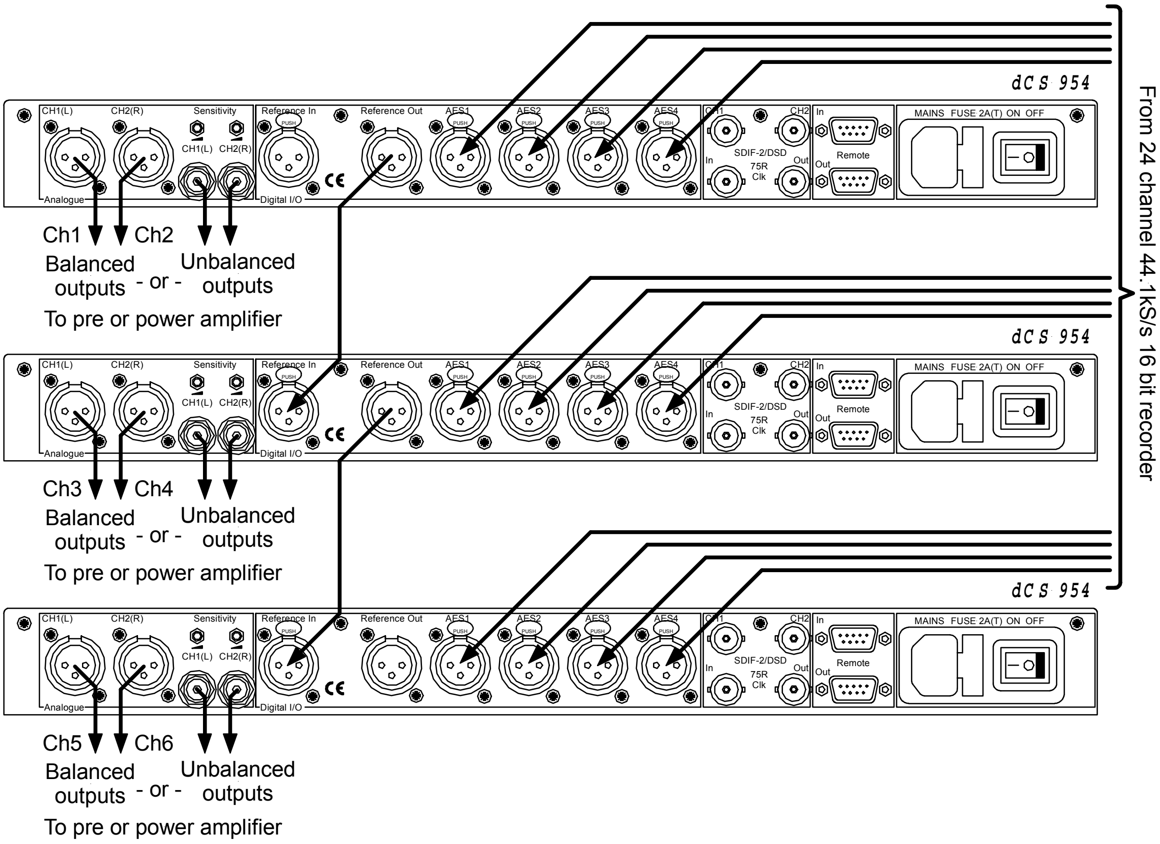

Figure 12 – Replaying a 6 channel DSD recording from a 24 track 16/44.1 recorder

do this: The top ***dCS 954*** needs to have its Ref In option set to Route, the middle one to Loop and the bottom one to Loop.t.

do this: Set all 3 units to DSD mode and press **AES 1** to select the 4-wire DSD input.

do this: Take care to ensure the input cables are connected in the right order.

The units self align quite accurately (see section **"Sample Alignment"** on page 41 onwards). Alternatively, Word Clock may be used as the syncing method, with no special set ups.

The source could be three 8-track recorders slaved together. The data streams must be bit-sync'ed.

Replaying 8 Channel P3D DSD

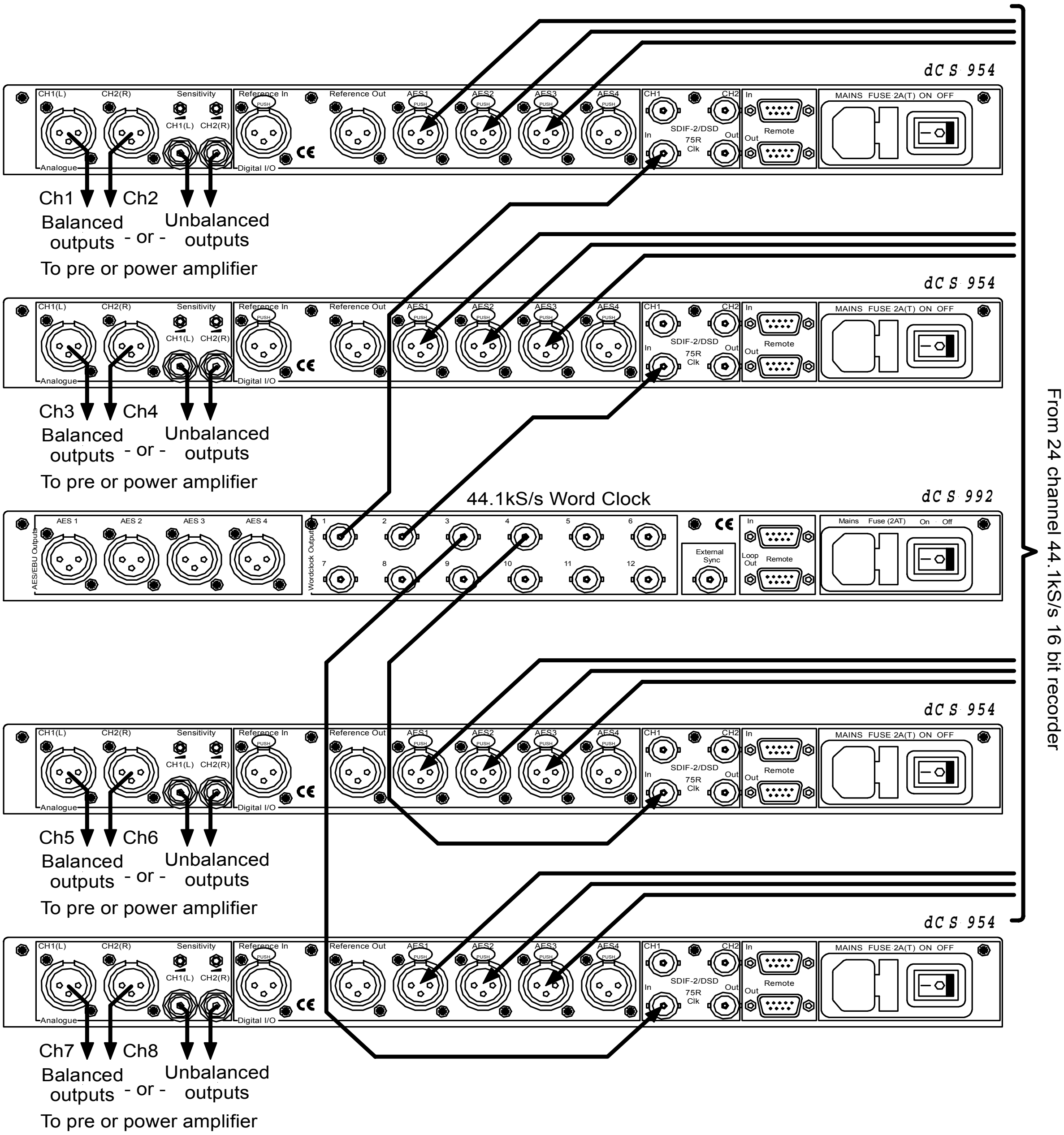

Figure 13 – 8 channel P3D DSD setup

Four P3D capable units and a Master Clock can be used as above to convert 8 channels of P3D encoded DSD.

do this: Set all 4 DACs to dSd3 mode on the DSD menu page and press the **AES1** button to select the P3D input.

do this: Select a filter and ensure **Mute** is off.

Upsampling a CD

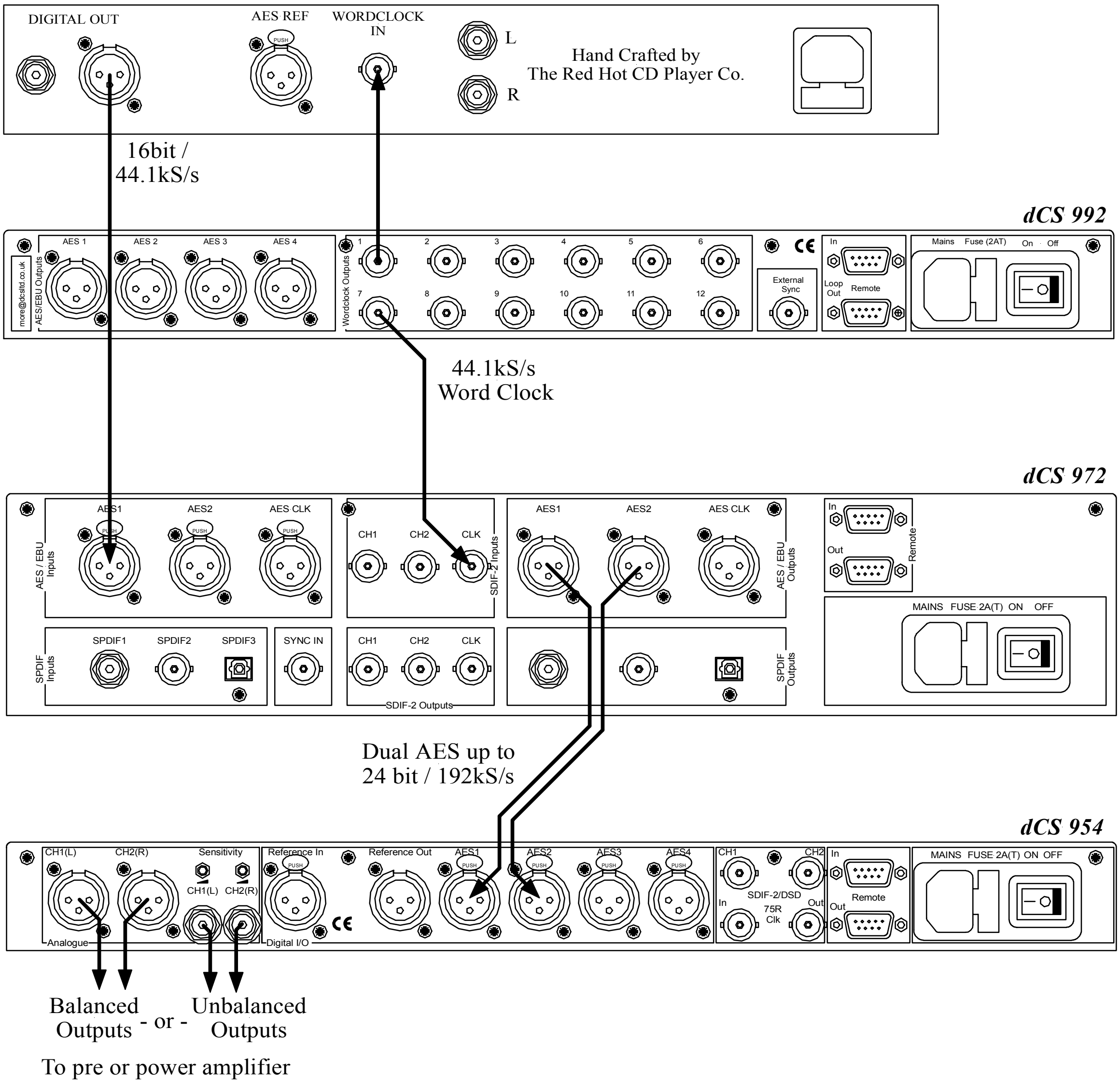

Figure 14 – Upsampling a CD to 24 bit / 192kS/s

do this: Set the ***dCS 972*** Input Frequency to Auto, Output Frequency to 192kS/s, Output Mode to Dual AES, Sync Source to Word Clock.

do this: Set the ***dCS 954*** to Dual AES on **AES1** & **2**, **Mute** to **Off** and select a filter.

IMPORTANT! *DO NOT lock the **dCS 954** to the **dCS 992** or the audio output will be full scale noise!*

The Master Clock is optional – it helps reduce jitter. If you don't have a Master Clock, you can slave the ***dCS 972*** to the CD player by setting Sync Source to Audio Input.

Converting Quad AES to Single AES

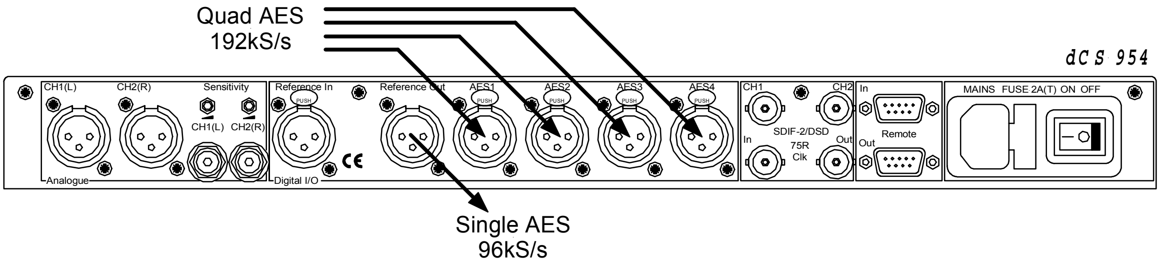

Figure 15 – Converting Quad AES to Double speed Single AES

do this: Select Quad AES and set the Ref In menu item to ddC.

IMPORTANT! *While the unit is performing a sample rate conversion, the analogue outputs are muted.*

The unit will also convert 176.4kS/s to 88.2kS/s.

Converting 4-wire DSD to SDIF DSD

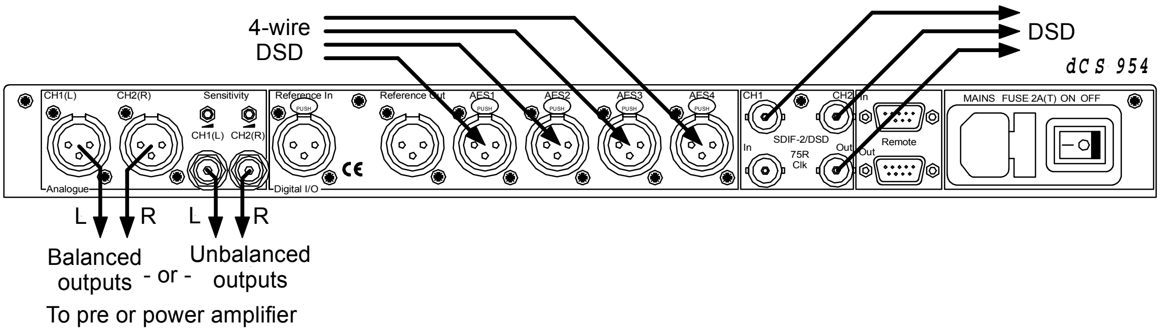

Figure 16 – Converting 4-wire DSD to SDIF DSD

do this: Set the dSd menu item to On and press **AES1** to select 4-wire DSD operation.

do this: Set the Ref In menu item to ddC.

do this: Select a filter and ensure **Mute** is off.

Replaying 24/192 from 2 Nagra-D recorders

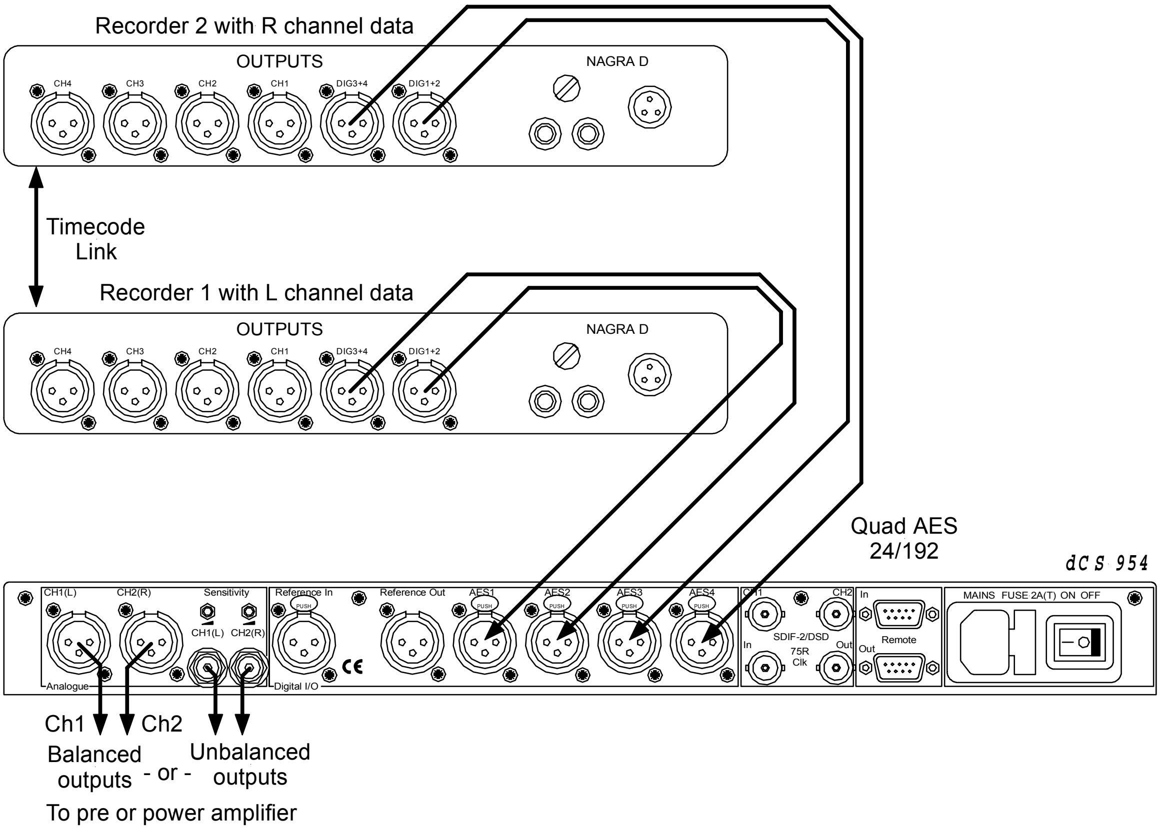

Figure 17 – Replaying 24/192 from 2 Nagra-D recorders.

Quad AES 24/192 recordings split between 2 tapes may be played back as shown. The 2 Nagra-D recorders must be fitted with special hardware and software to achieve this – contact Nagra for details. The ***dCS 954*** allows for small timing discrepancies between the AES1/2 pair and the AES3/4 pair.

dCS 954 TECHNICAL INFORMATION

Anti Image Filtering

The *dCS 954* offers a choice of 4 anti-image filters on most sample rates. These filters affect the ultrasonic part of the spectrum - 20 kHz upwards.

The unit is a DAC, with an output data rate set by the interface standard used. The bandwidth of the oversampling converter used is high, and so any signals that are in the input data will produce Nyquist images[4] in the output signal if they are not removed by filtering. The demands on this anti-image filter can be quite severe at the lower ("normal") sample rates - it must pass signals in the audio band (0 - 20 kHz) unimpaired, but it must remove images above Fs/2. This can result in a very sharp filter, and it is an unavoidable mathematical result that sharp filters have a poor, ringing, transient response. One effect of the ringing is to spread the energy in a transient over a significant period of time (it can be up to 1 ms). This seems to affect the stereo image that the ear would otherwise form.

One can trade off filter roll-off and energy smear - more relaxed roll-off gives less energy smear, but it may allow some of the signals in the input data to form audible images in the output signal. A signal containing Nyquist images can be corrected only by applying a sharp low-pass filter. However, as far as the ear is concerned, this may not matter. The ear can tell the frequency of a signal - up to a point. As the frequency rises, the accuracy with which the ear can tell what the frequency is decreases, and above a limit, all the ear can tell is that there is a signal, and it is above ... kHz. It can tell no more. So - it may be that a small amount of Nyquist imaging is acceptable to the ear.

The filters that we have included give increasingly good energy smear performance, and consequently have increasingly relaxed roll off. FiLT1 gives the sharpest roll off, with no Nyquist images, but the worst energy smear. Then as the number increases the smear decreases, but the imaging increases. Try them, to see which you prefer.

You may find that for different material, different filters are appropriate - and you may find that for different stages in the recording and mastering process, different filters are appropriate.

Our users tell us that they find the ability to select different anti-image filters very useful. Generally for classical music, number 2 is preferred while 3 & 4 are popular with users recording rock and jazz. Opinions differ widely – so try them for yourself.

The *dCS 954* 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"). The passband may or may not have a ripple[5], depending on the filter being used. The stop band is typically below –110dB0 and can be as low as –130dB0.

[4] See, for example "Principles of Digital Audio", 3rd Edition, by Ken C Pohlmann (McGraw-Hill Inc, 1995)
[5] Filters always have some ripple. For "zero ripple" filters this is in the µdB to pdB region.

Clocking

The sample clock quality significantly determines the performance of a DAC.

The highest quality clocks that are available are crystals, so we use these. The ***dCS 954*** uses one of two on-board voltage controlled crystal oscillators (VCXOs) as a clock source – one for 48 kS/s related outputs and one for 44.1 kS/s related outputs. When the unit is slaved to an external source, the appropriate VCXO is selected and synchronised to this by a phase locked loop (PLL). The PLL is of a special narrow bandwidth type, that provides a high degree of "clock cleaning" - but even so, signal quality may degrade if particularly poor source 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.

Synchronising to source

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

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.

There is a further discussion of some types of timing error in section **"Jitter and PLL bandwidths"** on page 68.

DSD

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 100kHz) although at these high frequencies, much noise is also present.

The SACD format sets 0 $dB0_{DSD}$ to be 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 range at the production stage do not move down into the audio band.

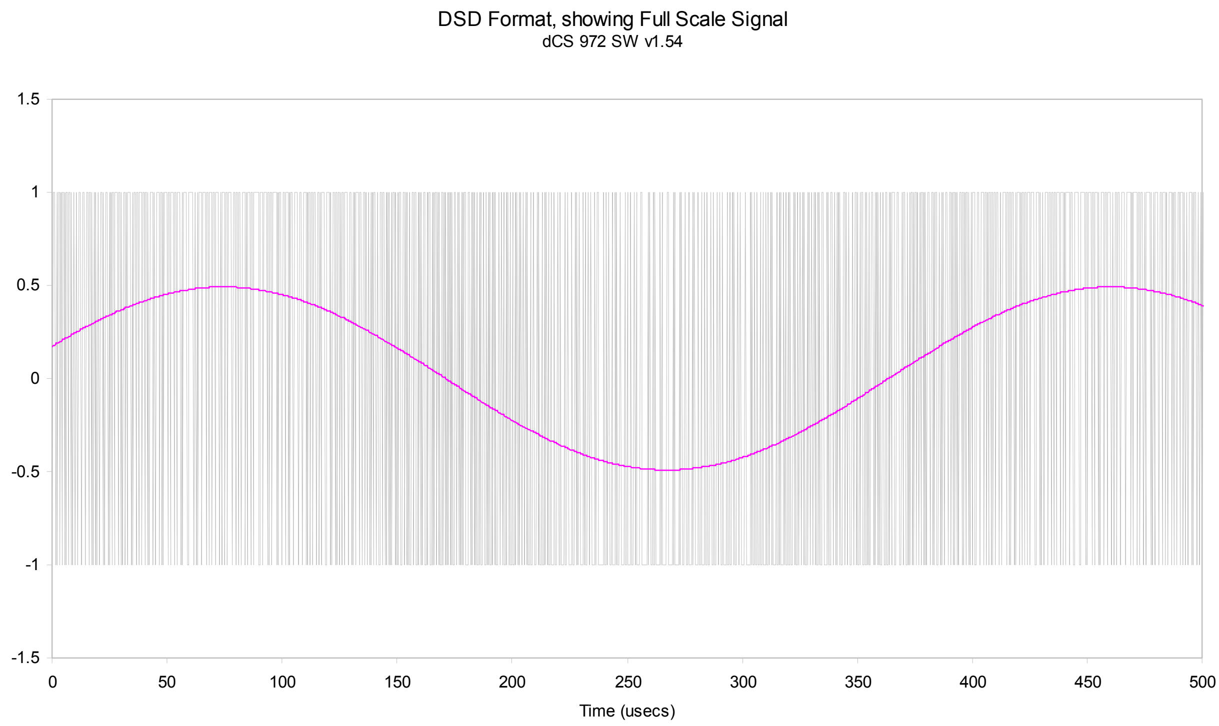

Figure 18 – DSD, showing DSD full scale

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

Electrically, TTL levels are used. There is no framing or block structure, and each channel uses one BNC connector. The Word Clock uses the third connector. See “**DSD on SDIF-2**” on page 54.

Sample Alignment

The ***dCS 954*** aligns samples such that Word Clock Out aligns with AES3 samples out (Reference Out), the rising edge of Word Clock Out 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.

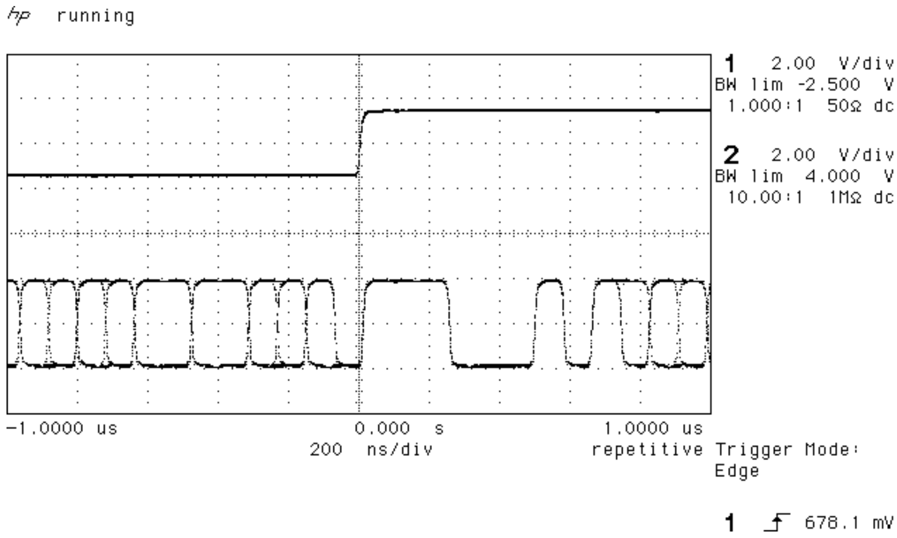

Figure 19 – Word Clock and AES3 outputs, 96 kS/s

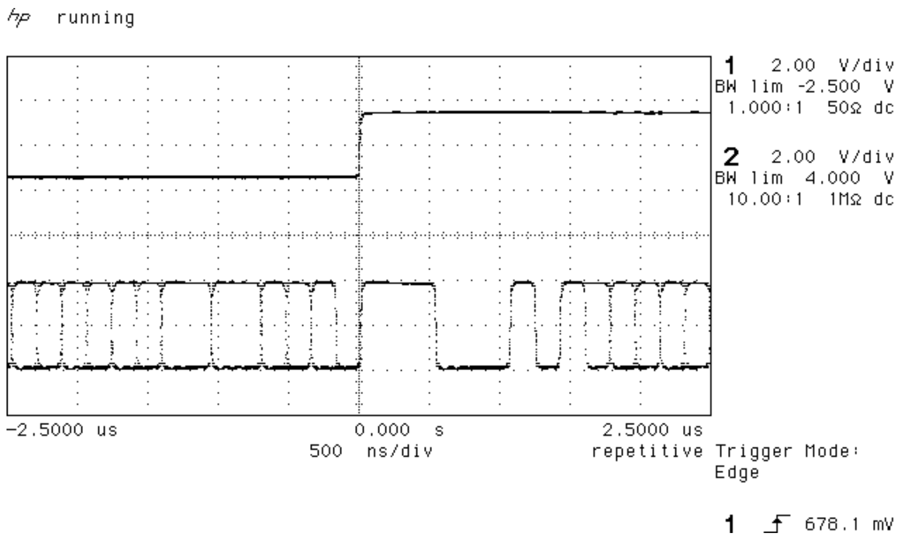

Figure 20 – Word Clock and AES3 outputs, 44.1 kS/s

When Word Clock 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. The misalignment is less than about 40ns. The scope shots below were taken with the unit sync'd to Word Clock In.

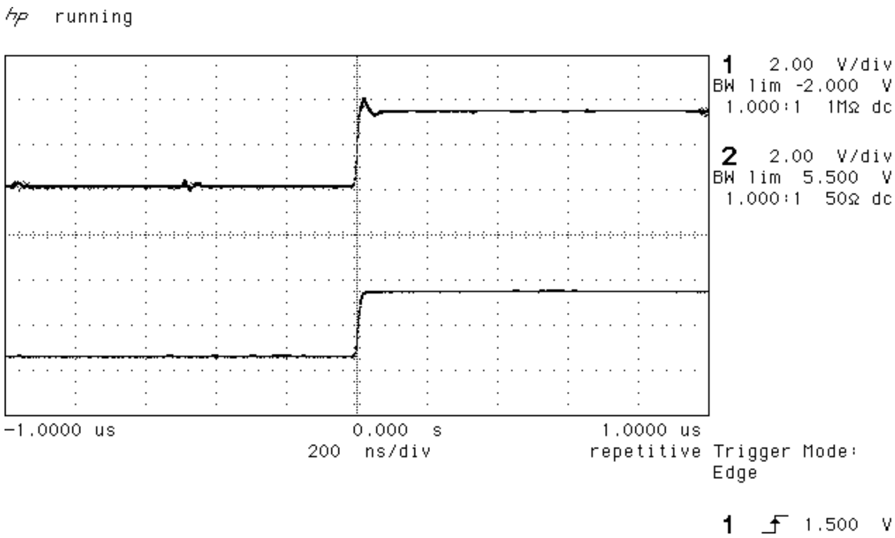

Figure 21 – Word Clock In to Word Clock Out, 96 kS/s

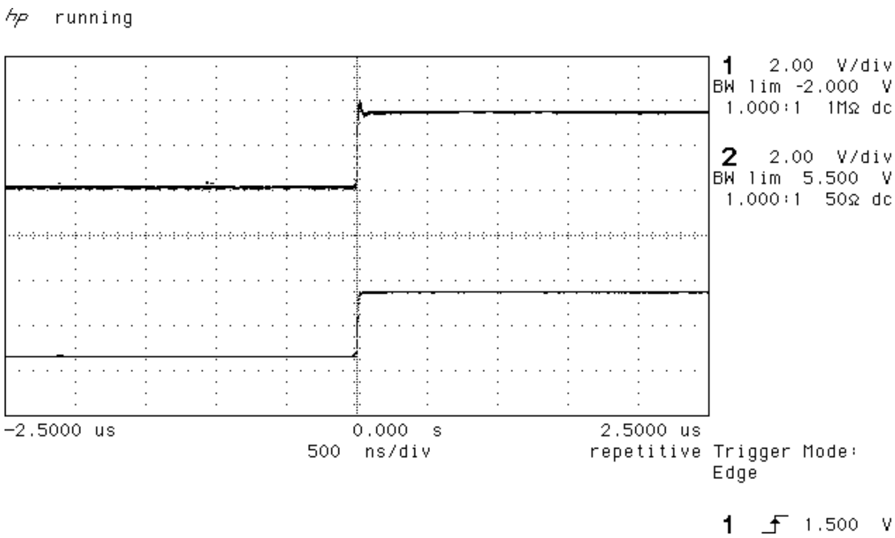

Figure 22 – Word Clock in to Word Clock Out, 44.1 kS/s

AES3 in and out (Reference 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 954***, the unit was slaved to **AES1** and the Ref In menu item was set to Route.

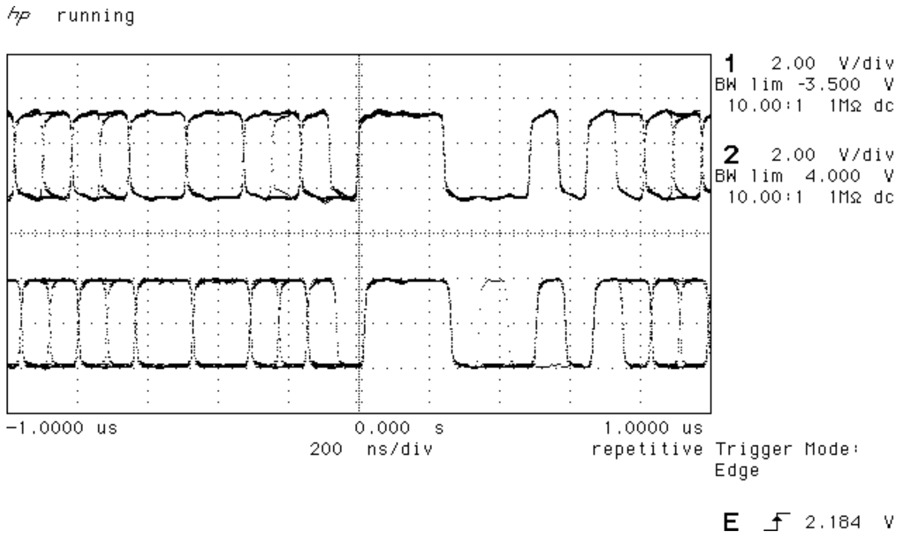

Figure 23 – AES3 in to AES3 out, 96 kS/s

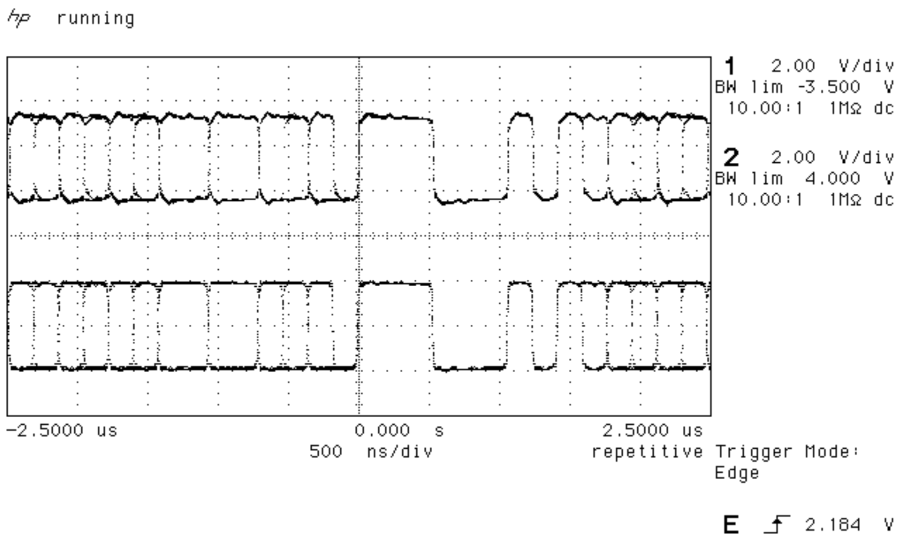

Figure 24 – AES3 in to AES3 out, 44.1 kS/s

AES3 Reference Out is also related to the phase of Clk In. The scope shots below were taken with the unit sync'd to Clk In

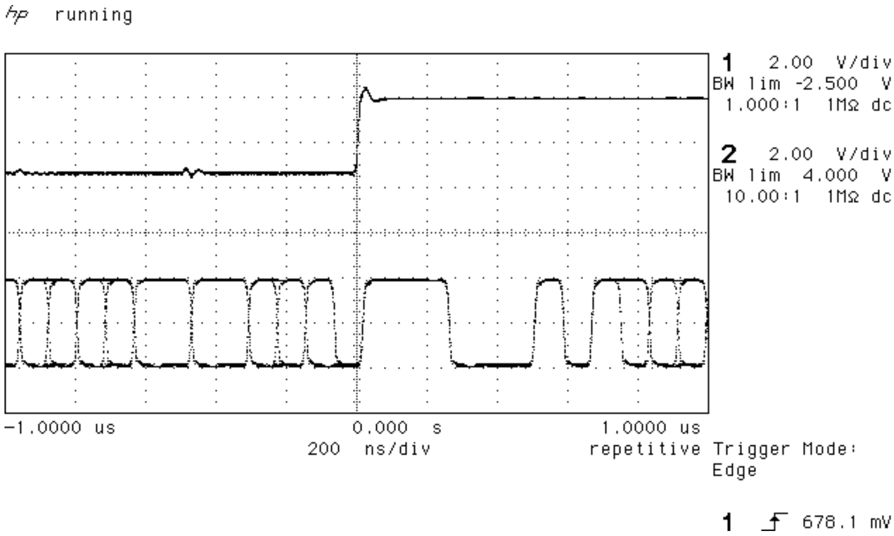

Figure 25 – Word Clock In to AES3 Reference Out, 96 kS/s

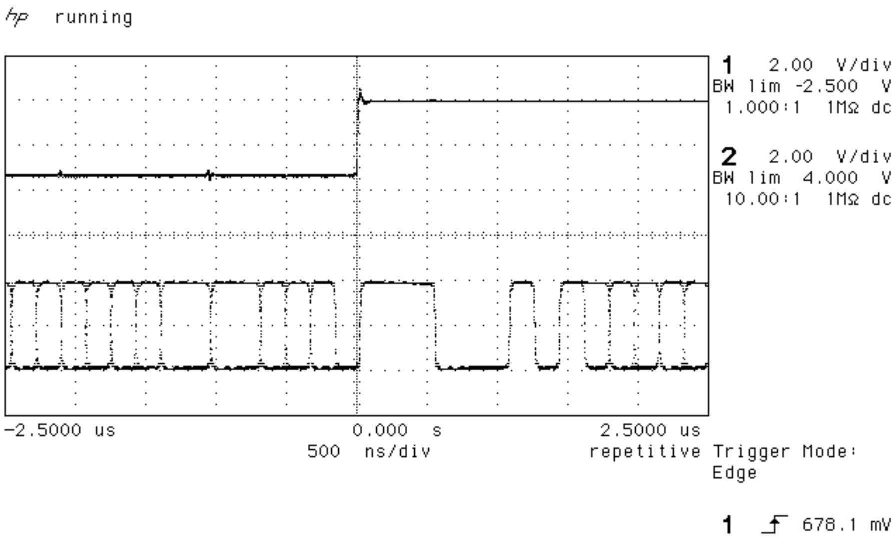

Figure 26 – Word Clock in to AES3 Reference Out, 44.1 kS/s

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
Damage level		> 20		V pk-pk
Connector		XLR3 female	XLR3 male	
Connections	Pin 1	Ground or shield		
	Pin 2	+Signal		
	Pin 3	-Signal		

Table 4 – AES/EBU i/o specifications

SDIF-2, SDIF-3 and DSD	Input	Output	
Type	*Single ended, ground referred*		
Impedance	100	25	Ω
Sensitivity (unloaded)	TTL	TTL	
Maximum Wordlength	24	24	bits
Damage level	> 10		V pk-pk
Time skew			
Word Clock in / out	< 40		ns
Connector	BNC x 3	BNC x 1	
Connections	CH1 (left)		
	CH2 (right)		
	Clk In & Out		

Table 5 – SDIF-2, SDIF-3 and DSD i/o specifications

Remote control interface	Input / Output
Type	RS-232
Level	RS-232
Baud Rates	1200, 2400, 4800, 9600
Data Format	Contact ***dCS***
Connector	9 way D type male

Table 6 – Remote Control Interface Details

Analogue Interface Specifications

Balanced Outputs			
Type		Balanced, semi-floating	
Format		AES14 : 1992	
Source Impedance (20Hz - 20kHz)		< 3	Ω
Maximum Load		600	Ω
Noise, unweighted (20Hz – 20kHz)		< -110	dB0
Spurious responses (20Hz - 20kHz)		< -100	dB0
		-110	dB0, typ
Signal Balance @	1kHz	> 40	dB, spec
	50Hz	45	dB, typ
	1kHz	53	dB, typ
	20kHz	53	dB, typ
L – R crosstalk (20Hz- 20kHz)		< -100	dB0
Level for Full Scale (as shipped)		+14	dBu
Trim range		±6	dB
Connector type		XLR3 male	
Connections	Pin 1	Ground or shield	
	Pin 2	+Signal	
	Pin 3	-Signal	

Table 7 – Balanced Output Details

Unbalanced Outputs		
Type	Unbalanced, ground referred	
Source Impedance	52	Ω
Maximum Load	600	Ω
Noise, unweighted (20Hz – 20kHz)	< -110	dB0, spec
Spurious responses (20Hz - 20kHz)	< -100	dB0
Level for Full Scale	+8	dBu
Connector type	RCA phono	

Table 8 – Unbalanced Output Details

Frequency Response

The overall frequency response is determined by the sample rate, the digital filter and the analogue filter. If imaging is to be avoided, all filters must cut-off before Fs/2 is reached, with a margin to allow for sufficient attenuation to be reached to effectively eliminate Nyquist images.

At a sample rate of 44.1kS/s, a flat 20kHz pass band must be maintained with at least 80dB of attenuation above Fs/2 (22.05kHz). This allows barely 2kHz margin between the pass band and the stop band for the filter to do its work, necessitating a very sharp filter.

At sample rates of 88.2kS/s and above, there is so much extra bandwidth available that maintaining a flat 0 - 20kHz audio band is less of a problem. This allows more relaxed filters to be used, resulting in extra sonic benefits.

Frequency responses for all 7 PCM sample rates, set to Filt1, are shown below.

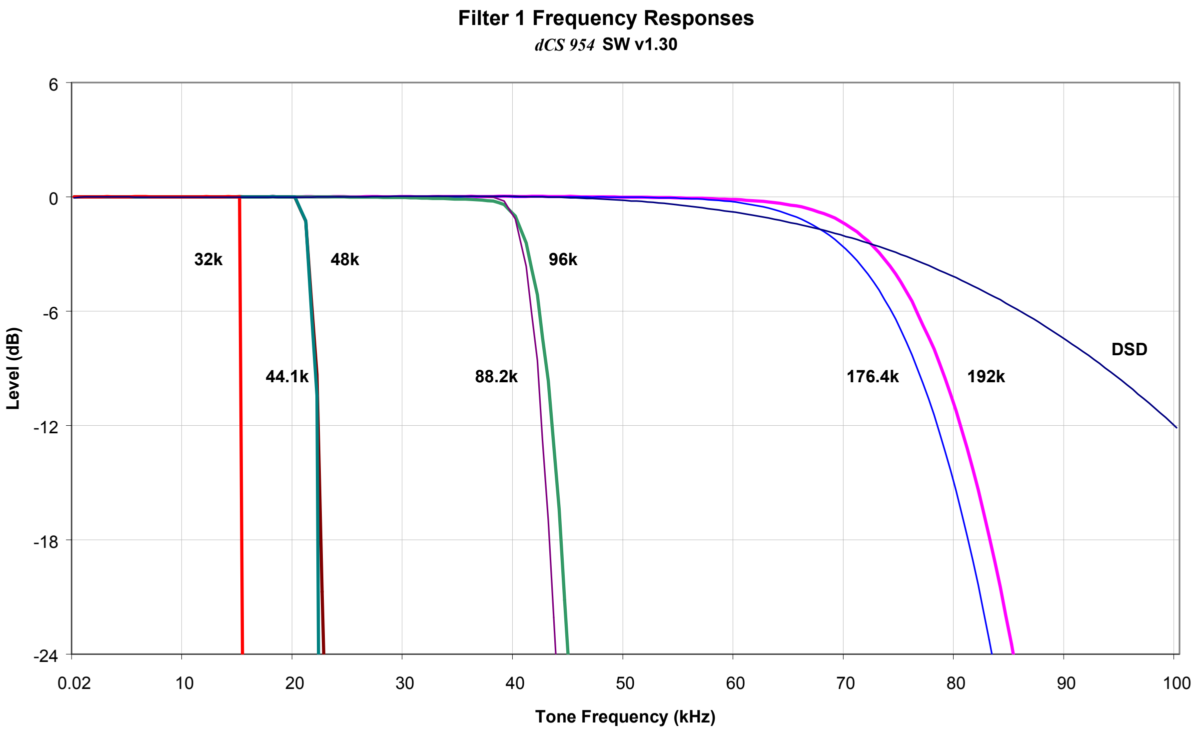

Figure 27 – Filter 1 frequency responses

Group Delay

The group delay for a ***dCS 904*** and ***dCS 954*** ADC and DAC were measured, at different sample rates. The results were as below (they are valid for software up to 1.5x):

	Filter 1	Filter 2	Filter 3	Filter 4
32kS/s	1378	1378	1378	1378
44.1kS/s	1335	1335	1335	1335
48kS/s	1258	1258	1258	1258
88.2kS/s	651	651	651	651
96kS/s	530	530	530	530
176.4kS/s	253	253	253	253
192kS/s	226	226	226	226
DSD	4	4	4	4

Table 9 – ***dCS 904*** ADC Group Delay in microsecs, v1.31 software

	Filter 1	Filter 2	Filter 3	Filter 4
32kS/s	1692	1710	1692	1700
44.1kS/s	1265	1262	1251	1245
48kS/s	1132	1136	1142	1150
88.2kS/s	517	238	267	191
96kS/s	414	210	224	168
176.4kS/s	129	129	129	129
192kS/s	119	119	119	119
DSD	23	23	23	23

Table 10 – ***dCS 954*** DAC group delay in microsecs, v1.30 software

The delays were measured analogue in to analogue out for a back to back pair. They are the same with SDIF-2 interfacing or AES interfacing. Once the pair had been measured, the DAC group delay was measured from SDIF-2 in to analogue out, and the ADC results inferred.

AES3 (AES/EBU) Format

Message Handling

The AES/EBU interface decodes 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

The message attached to the AES **Reference Out** depends on the Ref In setting. In Loop or Loop.t modes, it is copied from the **Reference In** data. When set to Route, the message is copied from the lowest numbered AES input selected. When set to ddC with a PCM input, the message is as follows:

Professional:	On
Emphasis:	Off
Non-Audio:	Off
Mode:	Not indicated
Sample rate:	(Correctly stated)
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[6], from the AES.

How Far will AES3 Go?

The AES/EBU format was designed to go reasonable distances, at 44.1 kS/s and 48 kS/s. Figure 28 and Figure 29 below show it over 16 m and 94 m using average cables. The waveform at 94 m can still be decoded, although it is quite degraded. Cable delay is about 5.6 ns/metre.

At 96 kS/s (twice the data rate the format was designed for) the allowed cable length is less. Figure 30 and Figure 31 below show this over 16 m and 94 m. At 16 m the waveforms are still very good, but at 94 m they are really quite unreliable.

We recommend restricting 96 kS/s cable runs to 20 m or less, and using good cable near this length.

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

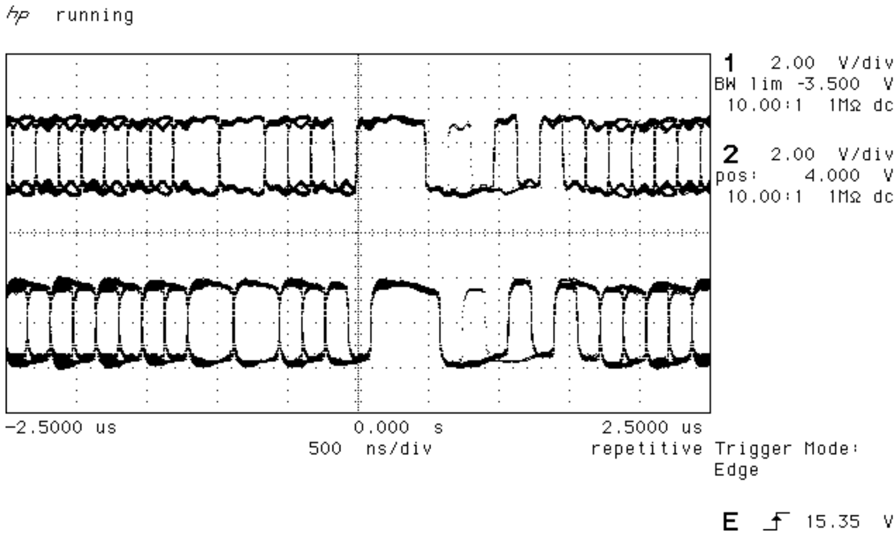

Figure 28 – AES3 format at 48 kS/s over 16 metres

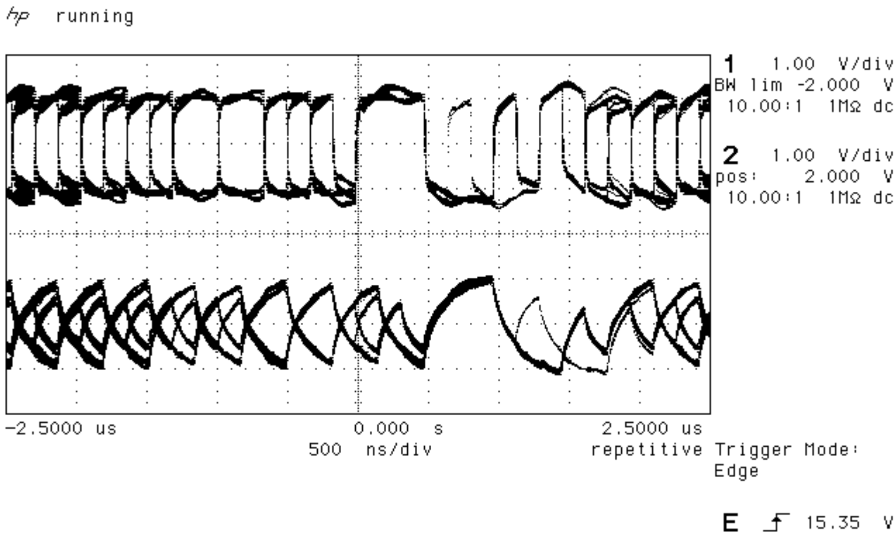

Figure 29 – AES3 format at 48 kS/s over 94 metres

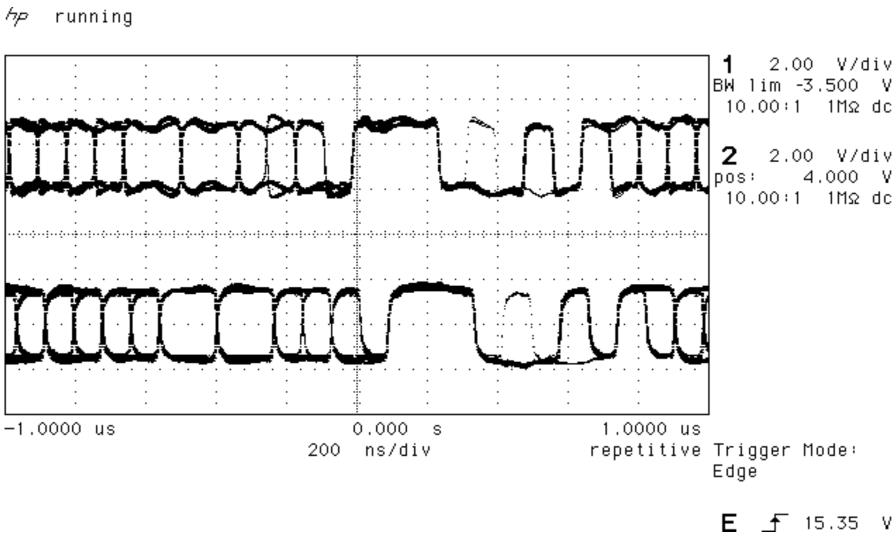

Figure 30 – AES3 format at 96 kS/s over 16 metres

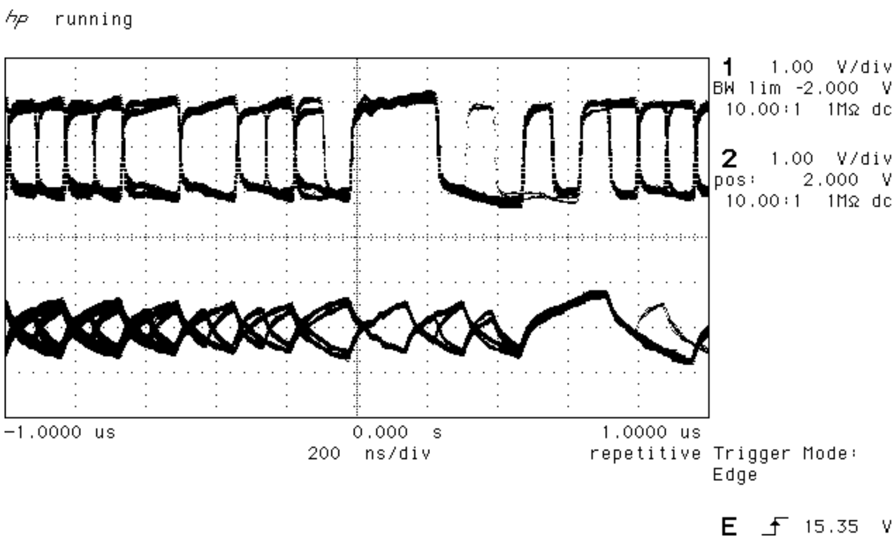

Figure 31 – AES3 format at 96 kS/s over 94 metres

SDIF-2

PCM Format

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 (Ch1)
Right Channel (Ch2)
Word Clock

The sync codes can 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 waveforms below show SDIF-2 waveforms (data and Word Clock) at 44.1 kS/s and 96 kS/s.

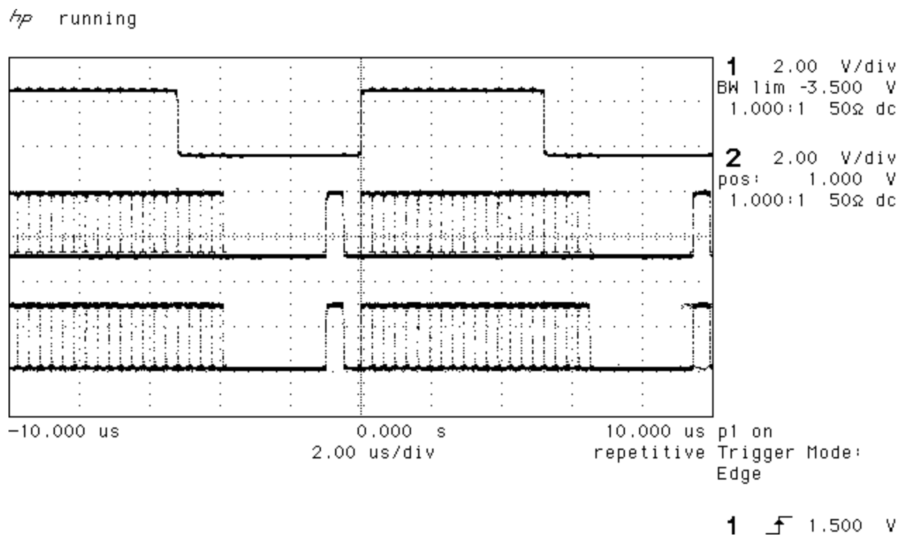

Figure 32 – SDIF-2 PCM format at 96 kS/s

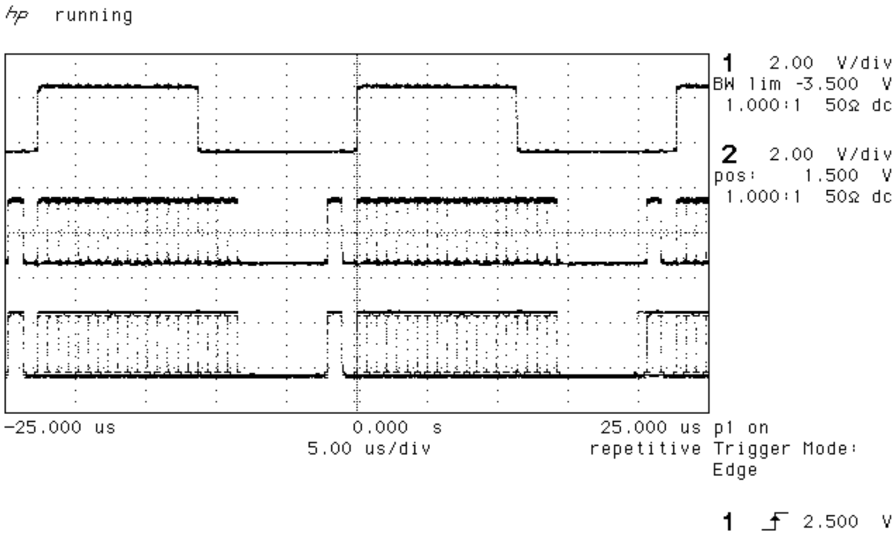

Figure 33 – SDIF-2 PCM format at 44.1 kS/s

SDIF-2 Messaging

The SDIF-2 message details are defined in the table following.

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

Table 11 – SDIF-2 Messages

DSD on SDIF-2

An SDIF-2 interface can be used for DSD. The waveforms appear quite different to PCM format. However, they do produce transitions where the illegal code transitions were, and for this reason we advise against locking to the illegal transitions in SDIF-2. We recommend always using Word Clock with SDIF-2 signalling.

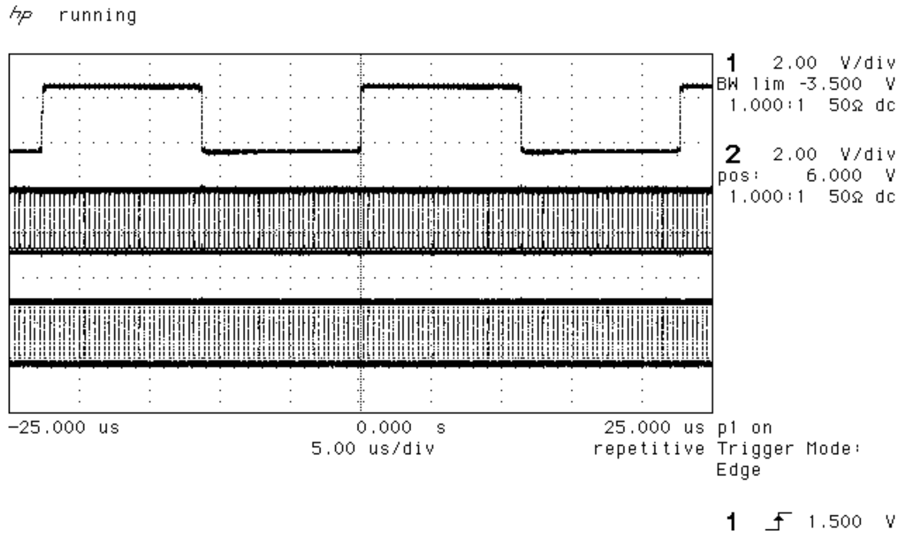

Figure 34 – DSD using SDIF-2 electrical format

DSD on SDIF-3

SDIF-3 embeds a clock in the SDIF-2 data stream, and so does not need word clock. It is used only for DSD – it is not used for PCM. Contact SONY for more details.

P3D Behaviour

Mute on CRC Error

The P3D format includes error checking. Data is packed into AES3 subframes, and each subframe is checked for integrity before it is converted to analogue. In the event of an error being detected in a subframe, the unit mutes the entire subframe, and passes control over to a more complex control mechanism. This looks for a long run of error free subframes before it unmutes – v1.36 software looks for 150 msecs – and displays the message “CRC”. A single data error can cause quite long mutes.

The reason for the presence of this test is to detect input data of the wrong format, which will cause full scale white noise, and can potentially damage equipment. Statistically, the wrong format data can cause a few subframes to pass the error check occasionally, so muting a single subframe is quite suspect.

Bit Error Rates

The bit error rate (BER) for a dCS 904/dCS 954 running P3D is below $5x10^{-14}$, (0 errors over 64 hours). The same error rate is likely to hold for all AES3 type formats.

RS-232 Remote Control Interface

Overall Description

dCS 9xx units can be controlled using a simple serial protocol, via the RS-232 ports, using the control format described below. All commands available from the front panel (and a few others, *dCS* use only) of a unit can be remotely controlled using this approach. Each unit must have a unique ID (in the range 0 to 99) which must be set up by hand using the menu system on the front panel. The units remember their ID when powered down, so this setting up only has to be done once.

Physical Interface

Units are all connected in a RS-232 daisy chain, up to a maximum of 11 units, with a serial cable (DB9 pin female straight cable) connected between the Serial Out and Serial In ports of the units. The same type of cable is used to connect the Serial In port of the first unit on the chain to the COM port of a PC.

By default all units are configured to operate at 1200 baud. Standard RS-232 signal levels are used. Bytes are transmitted with 1 start bit, 8 message bits, 1 stop bit and no parity.

Units can be switched to 1200, 2400, 4800 or 9600 baud. An RS-232 break will reset all units on the daisy chain to 1200 baud. A special command and ID is used to configure the units to other baud rates (see “Special Commands and Protocols” below). The following rates are recommended:

3 or less units	4800 baud
4 to 7 units	2400 baud
8 to 11 units	1200 baud

Operation of the daisy chain at higher than the recommended rates may result in incorrect behaviour of the system – either because the units misinterpret commands, or, more likely, because the controlling computer misinterprets their replies. Units will revert back to 1200 baud if they are switched off and on - they do not remember what they were last set to.

9600 baud is currently not fully tested over all temperatures. It can be used for single units operating in a benign environment.

Timing Accuracy and Warnings

The units use clock dividers derived from their crystals to produce the RS-232 signals. The frequency of operation is measured to be better than 2% for all baud rates with both crystals. Some of the commands, however, switch clock frequencies, and these may be controlled by phase locked loops with long time constants. While this is happening, correct RS-232 timing cannot be guaranteed, and the units should not be addressed – a period of 30 secs should be allowed after switching clock frequencies for timings to stabilise.

Units acknowledge and repeat back their actions on receipt of a command. The acknowledge should be waited for and checked before proceeding to the next command – see Acknowledge Message below

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

Transmit Message

The system employs the following protocol - all transactions are initiated by the PC. The PC is the transmitter and units on the daisy chain are receivers.

Byte 1 : ID of unit to process command
Byte 2 : Command (single byte)
Byte 3 : Length of parameter string
Byte 4 : List of parameters
.
.
Last byte : Checksum of parameter list

The minimum length of a transmit message is 4 bytes, maximum 64 bytes, limited by a buffer within the receiving unit. There are two cases for parameter length mismatch.

(a) Receiver expects n parameters, transmitter sends more.

This condition may arise when a later version remote tries to communicate with an earlier version of receiver. The receiver acts on first n parameters in list, ignores rest. The full parameter list is check summed.

(b) Receiver expects n parameters, transmitter sends less.

This condition may arise when a earlier version remote tries to communicate with an later version of receiver. The receiver acts on all parameters in list, and uses sensible defaults for the rest (ideally, no action except where this is silly). Do not truncate a command sequence expecting the receiver to do something sensible. The full parameter list is check summed.

The checksum is the sum of the bytes in the parameter list (bytes 4 to (last-1) byte) modulo 256. The receiving unit checks the checksum and will only act upon a command if the command is complete and the checksum is correct.

Acknowledge Message

The addressed receiving unit (ADC, DAC, DDC, Master Clock, etc) acknowledges within 50 msecs of the last transmitted byte in the transmit message. For some special cases (dCS use only, see "Special Commands and Protocols" below) commands do not acknowledge. If the checksum is incorrect the receiving unit will ignore the command, clear its buffer and will not acknowledge. Only valid command bytes will generate an acknowledge, other command bytes will be ignored, clearing the receiver buffer.

The acknowledge response starts with:

Byte 1 : 101010xx – indicates successful transmission to physical address, with xx indicating the time the unit may take to respond to the command

and then, with a command dependant part:

Byte 2 : ID of unit that processed command
Byte 3 : Command (single byte)
Byte 4 : Length of response string
Byte 5 : List of response bytes
.
.
Last byte : Checksum of response list

The checksum is the sum of the bytes in the parameter list (bytes 5 to (last-1) byte) modulo 256. The minimum length of an acknowledge message is 1 byte, maximum 64. If the checksum is incorrect the transmitter should re-issue the command.

For the first byte, the response times are:

xx
00 immediate (less than 50 msecs)
01 up to 3 seconds
10 up to 15 seconds
11 up to 25 seconds

The receiving unit will ignore any transactions on the RS-232 while it is busy. If the transmitter sends commands to a unit when the unit is busy the unit will not send an acknowledge back. The transmitter must be designed to time out after 50 msec and repeat the command if necessary. In a multi-unit environment, it would be sensible to organise the transmitter to access units with a "round robin" polling scheme – in this way several units can be instructed to perform commands simultaneously, the transmitter coming back to busy units periodically. It is also recommended that units are not accessed for the first ¾ of their "response" time – nothing untoward will happen, but the unit will be ignoring the RS-232 and will not respond, so the transmitter would just time out anyway.

Example :

To set unit 2 Emphasis to AUTO using the RS-232 control format:

transmit the string [2][34][1][0][0],
and the receiving unit will respond [169].

Special Commands and Protocols

BREAK

Continuous high on transmit line for more than 100 msec. Resets ALL units on daisy chain to 1200 baud.

GLOBAL ADDRESSES

Address F0 hex (240 decimal)

ALL units on daisy chain react to command. Nothing acknowledges. This should only be used for setting baud rates to 2400, 4800 or 9600 baud. Never change baud rate from a higher rate to a lower rate, as this could result in unexpected behaviour, always reset the daisy chain to 1200 baud and then issue the appropriate command. Never change the baud rate of a single unit in a multi-unit daisy chain as this could result in the chain locking up.

Address F1 hex (241 decimal), Command RS_ENABLE_DEBUG (19 decimal)

ALL units on daisy chain react to command. Nothing acknowledges. This enables dCS debugging commands. This may result in unstable behaviour of the unit.

Command Streams

Example – a system of 9 units with ID's set up as noted:

1 Master Clock (ID 1),
4 P3D compatible ADCs (ID 2, 3, 4 and 5),
4 P3D compatible DACs (ID 6, 7, 8 and 9).

RS232 operating at 1200 baud.

It is assumed that the transmitter operates on a round robin polling scheme and that each step completes before the next allowing for time outs. Except in the case of a time out a unit should not be accessed within the response time of its previous command. Within each step there is no need to wait for the command response time prior to moving on to the next unit – once an acknowledge has been received, the controller can safely assume that the unit is getting on with the command it has received, and can move on to the next unit. At the end of a step there is no need to wait before moving on to the next step.

Command strings are not given fully, the parameter string and the checksum are not explicitly given. A typical command is shown as:

[ID][Command Type], information about command

A typical response is:

[ACK Type][ID], information (when requested)

When changing the operating frequency of a unit the internal crystals are switched. It is recommended that after a crystal switch units are allowed to settle for a short time (< 1 second) to ensure optimum performance. In this case the units are being controlled by a Master Clock, so time should be allowed for this to switch and for the other units connected to it to also switch and begin to settle. It is recommended that there is no RS-232 activity for 3 seconds after the Master Clock frequency is switched to ensure all units have time to settle.

When operating in DSD mode units assume their reference clocks are operating at 44.1kHz. If a different frequency reference is used they will continuously monitor the reference clock frequency, preventing RS-232 accesses. It is therefore important to ensure the reference clock is set to 44.1kHz prior to entering DSD mode, and that DSD mode is left prior to changing the reference clock to another frequency.

Example: Switching to 96k PCM

The following example covers the system of nine units, in two complex format changes. Change the ADC and DAC operating mode to PCM prior to changing the Master Clock frequency. Change the DAC operating mode prior to the ADC. When changing the Master Clock frequency the system should be allowed to settle to the new frequency before any further RS-232 activity.

1) Command DACs 6, 7, 8 and 9 to change mode, the units may take up to 15 seconds to complete this command (if the previous mode had been DSD the FPGAs need to be re-loaded, which takes time). There is no need to wait prior to moving on to step 2.

Transmit -> [6][DSD_MODE], to change mode to PCM of unit 6
Responds -> [ACK 15 seconds][6], requested mode
Transmit -> [7][DSD_MODE], to change mode to PCM of unit 7

Responds -> [ACK 15 seconds][7], requested mode
Transmit -> [8][DSD_MODE], to change mode to PCM of unit 8
Responds -> [ACK 15 seconds][8], requested mode
Transmit -> [9][DSD_MODE], to change mode to PCM of unit 9
Responds -> [ACK 15 seconds][9], requested mode

2) Command ADCs 2, 3, 4 and 5 to change mode, the units may take up to 15 seconds to complete this command (if the previous mode had been DSD the FPGAs need to be re-loaded, which takes time). There is no need to wait prior to moving on to step 3

Transmit -> [2][DSD_MODE], to change mode to PCM of unit 2
Responds -> [ACK 15 seconds][2], requested mode
Transmit -> [3][DSD_MODE], to change mode to PCM of unit 3
Responds -> [ACK 15 seconds][3], requested mode
Transmit -> [4][DSD_MODE], to change mode to PCM of unit 4
Responds -> [ACK 15 seconds][4], requested mode
Transmit -> [5][DSD_MODE], to change mode to PCM of unit 5
Responds -> [ACK 15 seconds][5], requested mode

3) Check DACs for mode change. This command allows the Transmitter to check the mode of the DACs. If a unit has not changed the transmitter should go back to step 1 and repeat the command.

Transmit -> [6][REQUEST_DSD_MODE]
Response -> [ACK immediate][6], actual mode
Transmit -> [7][REQUEST_DSD_MODE]
Response -> [ACK immediate][7], actual mode
Transmit -> [8][REQUEST_DSD_MODE]
Response -> [ACK immediate][8], actual mode
Transmit -> [9][REQUEST_DSD_MODE]
Response -> [ACK immediate][9], actual mode

4) Check ADCs for mode change. This command allows the Transmitter to check the mode of the ADCs. If a unit has not changed the transmitter should go back to step 2 and repeat the command

Transmit -> [2][REQUEST_DSD_MODE]
Response -> [ACK immediate][2], actual mode
Transmit -> [3][REQUEST_DSD_MODE]
Response -> [ACK immediate][3], actual mode
Transmit -> [4][REQUEST_DSD_MODE]
Response -> [ACK immediate][4], actual mode
Transmit -> [5][REQUEST_DSD_MODE]
Response -> [ACK immediate][5], actual mode

5) Command Master Clock to change frequency. Allow the system time to settle after this command with no RS232 activity, three seconds should be sufficient.

Transmit -> [SEL_FS], change to 96k
Responds -> [ACK 3 seconds], requested frequency
Wait for 3 seconds

Check Master Clock has changed frequency. If it has not go back to step 5.

Transmit -> [1][REQUEST_FS], request actual frequency
Responds -> [ACK immediate][1], actual frequency

The system is now set up with the Master Clock configured for 96k operation and the ADCs and DACs locked in PCM mode to 96k.

Example: Switching to P3D

Change the Master Clock frequency to 44.1k prior to changing the ADC and DAC operating mode to DSD. Change the DAC operating mode prior to the ADC. When changing the Master Clock frequency the system should be allowed to settle to the new frequency before any further RS-232 activity.

6) Command Master Clock to change frequency. Allow the system time to settle after this command with no RS-232 activity, three seconds should be sufficient.

   Transmit -> [1][SEL_FS], change to 44.1k
   Responds -> [ACK 3 seconds][1], requested frequency
   Wait for 3 seconds

7) Check Master Clock has changed frequency. If it has not go back to step 6.

   Transmit -> [1][REQUEST_FS], request actual frequency
   Responds -> [ACK immediate][1], actual frequency

8) Command DACs 6, 7, 8 and 9 to change mode, the units may take up to 15 seconds to complete this command (if the previous mode had been PCM the FPGAs need to be re-loaded, which takes time). There is also no need to wait prior to moving on to step 9.

   Transmit -> [6][DSD_MODE], to change mode to P3D of unit 6
   Responds -> [ACK 15 seconds][6], requested mode
   Transmit -> [7][DSD_MODE], to change mode to P3D of unit 7
   Responds -> [ACK 15 seconds][7], requested mode
   Transmit -> [8][DSD_MODE], to change mode to P3D of unit 8
   Responds -> [ACK 15 seconds][8], requested mode
   Transmit -> [9][DSD_MODE], to change mode to P3D of unit 9
   Responds -> [ACK 15 seconds][9], requested mode

9) Command ADCs 2, 3, 4 and 5 to change mode, the units may take up to 15 seconds to complete this command (if the previous mode had been DSD the FPGAs need to be re-loaded, which takes time). There is also no need to wait prior to moving on to step 10.

   Transmit -> [2][DSD_MODE], to change mode to P3D of unit 2
   Responds -> [ACK 15 seconds][2], requested mode
   Transmit -> [3][DSD_MODE], to change mode to P3D of unit 3
   Responds -> [ACK 15 seconds][3], requested mode
   Transmit -> [4][DSD_MODE], to change mode to P3D of unit 4
   Responds -> [ACK 15 seconds][4], requested mode
   Transmit -> [5][DSD_MODE], to change mode to P3D of unit 5
   Responds -> [ACK 15 seconds][5], requested mode

10) Check DACs for mode change. This command allows the Transmitter to check the mode of the DACs. If a unit has not changed the transmitter should go back to step 8 and repeat the command.

   Transmit -> [6][REQUEST_DSD_MODE]
   Response -> [ACK immediate][6], actual mode
   Transmit -> [7][REQUEST_DSD_MODE]
   Response -> [ACK immediate][7], actual mode

Transmit -> [8][REQUEST_DSD_MODE]
Response -> [ACK immediate][8], actual mode
Transmit -> [9][REQUEST_DSD_MODE]
Response -> [ACK immediate][9], actual mode

11) Check ADCs for mode change. This command allows the Transmitter to check the mode of the ADCs. If a unit has not changed the transmitter should go back to step 9 and repeat the command.

Transmit -> [2][REQUEST_DSD_MODE]
Response -> [ACK immediate][2], actual mode
Transmit -> [3][REQUEST_DSD_MODE]
Response -> [ACK immediate][3], actual mode
Transmit -> [4][REQUEST_DSD_MODE]
Response -> [ACK immediate][4], actual mode
Transmit -> [5][REQUEST_DSD_MODE]
Response -> [ACK immediate][5], actual mode

The system should now be set up with the Master Clock configured for 44.1k operation and the ADCs and DACs locked in P3D mode.

Command name	Command Byte	Number of Parameters in Command	Parameters	Parameters in Response	ADC	DAC	DDC	MClk
RS_AUTO_SLAVE	15	1	0 = do not automatically slave 1 = automatically slave to a reference input	0	X			X
RS_MASTERSLAVE	16	2	First parameter 1 = Master 0 = Slave. If slave, second parameter: 0 = AES 2 = SDIF	0	X		X	X
RS_ENABLE_DEBUG	19	3	Global Command	None	X	X	X	X
RS_SEL_FS	32	1	Select Output Fs	Echos message	X	X	X	
RS_FILTER	33	1	Select Filter, 0-3	0	X	X	X	
RS_EMPH	34	1	Select De-emphasis filter to use, 0 = Auto 1 = 50/15 2 = CCITT 3 = None	0	X	X	X	
RS_OUT_MODE	36	1	0 = Output SDIF wordclock on w/clk out, 1 = Output AES on w/clk out	0	X	X		X
RS_TRUNC	39	1	No. of output bits (16 - 24)	0	X		X	X
RS_SNS	40	1	Noise shaper, 0 = Auto 1 = Off 2 = 1st Order 3 = 3rd order 4 = 9nth order	0	X		X	X
RS_DDC	41	1	0 = Normal mode (e.g. D in A out for DAC, A in D out for ADC), 1 = D in D out	0	X	X		X
RS_OUT_RATE	42	1	0 = Low speed output (e.g. Dual 88.2/ Quad 192 1 = High speed option	0	X		X	X
RS_MUTE	43	1	0 = Unmute 1 = Mute	0	X	X	X	X
RS_AUTO	44	1	1 = Turn off automatic input selection (DAC)	0		X		
RS_7SEGS	47	1	1 = Turn off 7-segment display	0	X	X		X
RS_INP_FORMAT	48	1	Select Input format for DACs – 0 = Auto 1 = Single wire 2 = Dual Wire 4 = Quad	0		X	X	
RS_4WIRE	49	1	0 = Enable 4-wire DSD outputs 1 = Disable 4-wire DSD outputs	0	X			
RS_FLIP	50	1	0 = Normal 1 = Flip channels	0		X	X	
RS_ACUT	51	1	1 = Disable Auto Digital Muting	0		X		
RS_FINE_LOCK_MODE	52	1	1 = Use coarse lock 0 = use fine lock	0		X		
RS_WAVETYPE	63	1	0 = Signal Generator Off 1 = Signal Generator ON	0	X	X	X	X
RS_AMP	64	1	Generator Amplitude, format X	0	X	X	X	X
RS_FREQ	65	1	Generator Frequency. Specified as a 32 bit number. Expressed as a fraction of Sample Frequency	0	X	X	X	X

Command name	Command Byte	Number of Parameters in Command	Parameters	Parameters in Response	ADC	DAC	DDC	MClk
RS_REF_MODES	77	2	first parameter is terminator (AES) 0 = unterminated 1 = terminated second is reference mode 1 = ref out is internal 0 = pass through	0	X	X		X
RS_OVLD_LEV	87	1	Overload threshold, format X	0	X			
RS_VOL	111	1	Digital volume control, format X	0		X	X	
RS_PHASE	112	1	Phase: 0 = None inverted 1 = Both inverted 2 = left inverted 3 = right inverted	0		X	X	
RS_REF_MODE	114	1	Select reference input to clock from: 0 = AES1 1 = AES2 2 = SDIF-2 Clock (word clock) 3 = SPDIF1 or AES3 4 = SPDIF2 or AES4 5 = SPDIF3	0		X	X	
RS_DSD_MODE	119	1	0 = DSD Off 1 = DSD (SDIF) 2 = 4-wire DSD	Echos message	X	X	X	
RS_BAUD_RATE	141	1	Global command	None	X	X	X	X
REQUEST_DSD_MODE	142	0	response -> DSD mode	Yes				
REQUEST_FREQUENCY	143	0	response -> Frequency of unit	Yes				

Table 12 – RS-232 Command Set

Format X – the level set number is –0.1dB times the 16 bit (positive integer) used. So, for example, 260 would set –26dB below full scale for generator amplitude.

Power Consumption

The ***dCS 954*** has a linear power supply, and so power consumption changes as the mains voltage changes. The internal regulation is comparatively efficient for a linear supply, so these changes are kept to a minimum. Power consumption is independent of mains voltage setting.

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

Nominal mains	28 W
Mains -10%	26 W
Mains +10%	31 W

The actual intended supply voltage is shown on the rear panel. 50 Hz or 60 Hz 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 70 in this manual and contact your distributor or ***dCS***.

Size and Weight

The ***dCS 954*** 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	44 mm	(2U)
Height, with feet	52 mm	
Depth	390 mm	see note (ii)
Weight	6.8 kg	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.
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 ***dCS 954*** has no ventilation slots or fan cooling. It dissipates relatively low power, so that usually allowing natural convection provides enough cooling in most circumstances. It is sensible, however, to not install the unit near heat sources such as radiators, hot air ducts or in direct strong sunlight.

Operating conditions should be such that internal temperature does not exceed 70°C substantially, as read out from the internal temperature sensor (see the menu function **Heat** on page **Error! Bookmark not defined.**). This will tend to be met if the ambient temperature is below 50°C, although it will depend a bit on how the unit is positioned. Internal temperature should not fall below 0°C, and should be a non-condensing. The unit monitors its internal temperature, and displays one of two error messages as the temperature rises. At and above an internal temperature of 78°C, the unit displays Hot on its front panel, as a warning. Performance and reliability will be degraded if operated in this range for long periods. At and above 88°C the unit displays Ouch, and should be turned off. See section **System Messages and Error Codes** on page 77.

Figure 35 below shows the rise of internal temperature for the middle unit of three stacked as in a rack, with support plates between. Allowing 3 cms between units gives reasonable cooling.

If in doubt, the easy test is – the ***dCS 954*** is happy to work anywhere a human is.

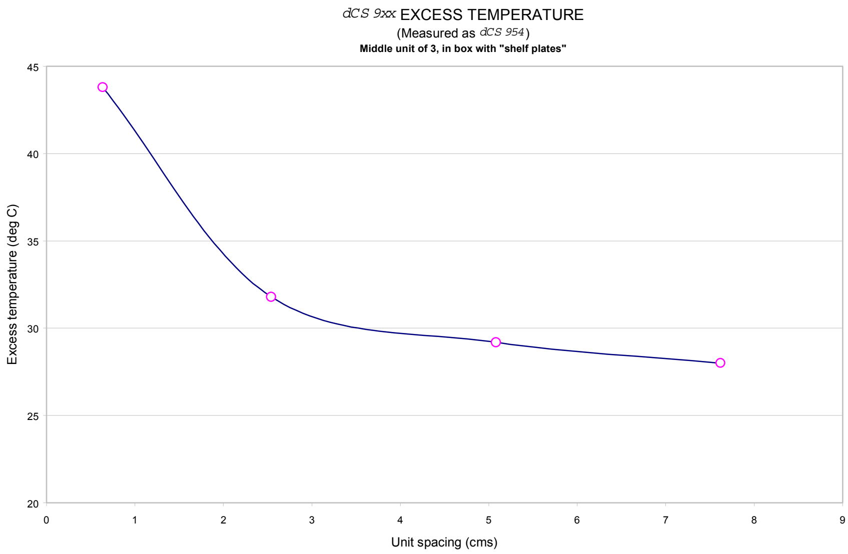

Figure 35 – Temperature rise above ambient for a unit in a stack of 3 with poor ventilation

General Technical Information

Jitter and PLL bandwidths

Jitter and PLL performance are related. In a DAC, in many applications the clock for the received data has to be extracted from the signal coming into it. To do this, the DAC has to have circuitry that looks at data edges (edges are the only things that carry time information), and has to extract enough information from these to generate an stable internal clock. This stability criterion is much greater for a DAC, which has to produce an analogue output, than for a piece of digital equipment – which just has to avoid data corruption.

The task is generally carried out in a phase locked loop (PLL). This controls an internal oscillator (a VCXO in ***dCS*** equipment) such that on average the rate that this produces clock edges is the same as the rate the incoming signal produces clock edges (the frequencies are the same), and such that the phases of these two clocks (incoming and internally generated) is on average fixed – they line up in the same way each time.

“On average” is the key phrase. The purpose of the PLL is to produce a clean clock from one that may have come through a lot of digital equipment, and may not be so good. So, the internal clock has to allow the incoming one to wander around a bit (jitter) without causing any local upsets. The rate that the internal one changes if the incoming one changes is related to the bandwidth of the PLL. If the bandwidth is high, the internal one tracks rapid changes in the incoming signal, and jitters the output in line with the input accordingly. If the bandwidth is low, the clock used for reconstituting the analogue signal can be very good and jitter free, but at any particular time the difference in phase between the two clocks can be substantial. This can cause decoding errors.

In principle, the lower the PLL bandwidth, the more the DAC clock can be made independent of the incoming clock, and so the more jitter can be removed. Two things conspire to limit how far one can go with this process.

The first is “lock in time” – the time it takes the low bandwidth PLL to lock to a new signal. As the bandwidth reduces this can get very long.

The second is low frequency jitter in the incoming signal. Most signal sources with reasonable clocks have noise and spuriae in the clock spectrum that increase as one gets closer to the clock frequency. As one gets very close, these cause large, slow time excursions – edges wander on a slow basis. At bandwidths in the Hz area, with sources that involve any form of mechanical device (storage drives, for example), these can be many hundreds of nsecs, and if one goes below 1 Hz, they get worse.

Because of these types of issue, dCS use a bandwidth of around 5 Hz for our PLLs in fine lock. This bandwidth enables jitter in the audio band to be substantially suppressed, but lock in times do not become excessive. We use a dual arrangement, with one low bandwidth PLL used to extract the clock (the low bandwidth one), and a much faster one used to extract the data. The bandwidth of the data extraction PLL has no effect on audio quality – as long as it extracts digitally correct data it is doing its job okay. It is capable of correctly extracting data with quite large time errors, easily meeting the AES3 requirements. Using this approach, rather than any approach based on FIFOs, ensures that delay between data coming in and replaying is minimised. If a FIFO approach is used,

the FIFO has to be significantly filled at all times, which is the same a delay in the signal path.

A bit error rate measurement based on this approach has shown rates with an upper bound of $5*10^{-14}$.

OPTIONS

Mains Voltage

We ship with the mains wired according to the destination. The voltage option should be specified when the unit is ordered, by specifying the country of use. It can be updated later by your dealer, if necessary.

Video Frequency VCXOs

We can fit additional video frequency VCXOs (enabling frequencies such as 44.056kS/s and 47.952kS/s). These are best fitted at ***dCS***, to allow full checking.

P3D, DSD Pro and Other Formats

We can fit larger FPGAs to allow P3D, DSD Pro and other formats. This has to be done at ***dCS***.

Ordering Options For A New Unit

To order any option, just tell us:

dCS 954 for use in <*country*>, with options ….

IMPORTANT! *Always specify the intended country of operation, otherwise we will assume that country of delivery is the same as country of operation.*

Having Your Options Changed

dCS support modifications, updates and option changes to supplied ***dCS 954*** units. If you are in any doubt, please contact your Distributor or ***dCS***. In general, these will be carried out at ***dCS***, because we have extensive test facilities and can verify the changes.

IMPORTANT! *Please do not attempt the changes yourself. The unit's performance and 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 date you purchased your unit. 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 (260 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 mains (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 guarantee. 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.

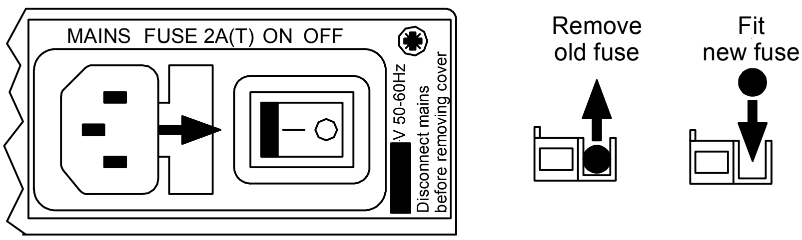

Figure 36 – Changing the Mains Fuse

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

Software

Installing New Software

Updated operating software can be downloaded via the RS-232 link from a PC comm. port, using the Windows Remote software running on the PC, or can be copied from an EPROM installed internally.

Using the RS-232 download is hands free, but takes about 40 mins per unit. With special software (contact dCS) multiple units can be daisy chained together so that one PC can update them all serially (overnight).

To update the software by the RS-232 link, load the new software into a convenient directory on the PC, then run the Windows 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 it thinks is connected, and then reports back. For each unit there is an **Info** button. Select the **Info** button for the unit you wish to update, and then select **Download Flash**. The programme will prompt you for the file to use, and then will start the download. If you want to programme many units automatically (say overnight) contact dCS for special software to enable this function.

IMPORTANT!

Do not turn the unit off until the download is complete. The unit has to erase its current programme before it can store the new one, so if the power is turned off, its internal programme store will have been erased but no new programme installed. Contact dCS if this happens inadvertently – the situation can be recovered if it does happen, but it involves taking the lid off the unit.

To find out if there are any software updates available for your equipment, call us, or email us, with your units serial number, or check our web site (www.dcsltd.co.uk). In general, software updates are free. Manuals for updated software can be downloaded from our web site, or just call us.

During An Update …

As soon as the download starts, the ADC will display **Prog**. The Windows programme will say **Erasing Flash** (10 secs), then **Flash Erased** (quick) then **Programming Flash**. At this stage a progress bar with a count down time is displayed, showing how much time is left (30 mins or so). After this has counted down, the PC says **Done** and the ADC reboots itself. Depending on the nature of the software update, it unit may then need to re-initialise its internals – if it does it will say **Hold** on its front panel. Do not do anything at this stage. Then, when that message disappears, it will be back to normal use.

Hardware Update or Calibration

You may wish to have your unit updated occasionally. ***dCS*** offer this service - we will install any modifications or hardware updates that have occurred since your unit was first shipped, and give the unit a full retest to current standards, including re-calibrating its VCXOs (which drift over time). 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.

Warranty

Your ***dCS 954*** is guaranteed for a period of 12 months against faulty workmanship or materials. Warranty repairs should only be carried out by ***dCS*** or an authorised distributor. This warranty will be invalidated if the unit is misused or tampered with in any way.

Safety and Electrical Safety

There are no user serviceable parts inside the ***dCS 954*** 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 954***.

TROUBLESHOOTING

Error Codes and Messages

The error codes and messages reported by ***dCS 954*** provide an effective means to diagnose the majority of problems that may be encountered in use - including problems with the overall system the unit operates in, internal device warnings and internal device failures. Please note that through damage or component failure, the unit self check may fail to operate. If this happens, please contact your distributor or ***dCS*** for assistance.

Internal Device Error Codes

Sometimes the unit may misbehave. If there is an internal reason, an internal device error code may be displayed as follows:

Err.xy an error xy (see table below) has been detected

where xy values have the following meanings:

Code	Description
01	E^2 memory (EEPROM) not present
02	Error initialising DSP
03	Error loading DSP
04	Error initialising DSP for coefficients
05	Error initialising DSP for coefficients
06	Error loading DSP coefficients
07	Error loading DSP coefficients
08	Error sending command
09	Error sending command
10	Error sending command
11	Error sending command
12	Error with LSB/MSB configuration
13	Error with LSB/MSB configuration
14	Error with LSB/MSB configuration
15	Error configuring FPGA
99	DSP error

Table 13 – Internal Error Codes

If you get any of these, please contact ***dCS***, with as much information as possible to help us re-create the problem. Some of these may have hardware problems as their cause, some may have software.

System Messages and Error Codes

Some other messages may be displayed that give indications of errors from other sources (outside the unit):

Display	Description
n.Aud	The data has been flagged by an AES3 message as Non Audio (perhaps a CD ROM). This message may also be displayed briefly when the sample rate is changed (see page 22)
Hot	The unit is overheating, and performance may suffer.(see page 66)
Ouch	The unit is seriously overheating, and may be damaged shortly. Switch off! (see page 66)
Bad Fs	The sample rate coming in is not one the unit can lock to, or there is an input signal quality problem.(see page 39)
CRC	P3D error message only, if the data coming in does not pass the CRC data integrity check (see page 55)

Table 14 – System Error Codes

Trouble Shooting Your System

If you experience difficulties when using your *dCS 954*, the following suggestions may help to resolve the problem.

The unit fails to power up

- Ensure there is power available on the mains cable and the unit's mains switch is On.
- Check the rated supply voltage shown on the rear of the unit matches the local supply voltage.
- Check that the fuse has not blown - if so, correct any obvious cause then replace the fuse as described in the section "**User Changeable Parts**", page 72.
- Check that the mains cable is pushed fully home into the mains inlet in the rear of the unit.

The unit fails to lock to a source

- Ensure the correct input is selected and the Ref In menu item is set to Route.
- Check that the digital audio cable is of the correct type, correctly connected and not damaged. Damaged cables are a VERY common cause of malfunctions!
- Check that the source is switched on.
- Some CD players do not generate a digital output unless the disc is playing - set the player in "play" mode and check that the unit locks.
- If you are using SDIF-2, set the BNC I menu item to SDIF and the BNC menu item to Input.

The unit fails to respond to the controls

- While locking to a source or changing some settings (e.g. Filter), the microcontroller inside the unit is busy and will not respond to new commands for a few seconds. Turning DSD mode on and off occupies the microcontroller for about 15 seconds.
- Short mains supply drop-outs may cause the microcontroller to lock up. Switch off the unit, wait 10 seconds then switch on again. If this does not clear the fault, please contact your dealer.
- The Remote Control disables the front panel controls.

The audio output is low or absent

- Check that all cables are connected correctly and not damaged. Damaged cables are a VERY common source of malfunctions!
- Check that the source and destination equipments are switched on and correctly set up.
- Check that the unit is locked to the source you want to monitor.
- Ensure **Mute** is not enabled - LED off.
- Set the **Volume** control 0.0dB.
- Ensure the source is sending audio data.
- If the Ref In menu item is set to ddC, change it to Route. With PCM inputs, DDC mode disables the analogue outputs.

The level trimmers on the rear panel do not change the output level

- Ensure the trim tool or flat-bladed screwdriver you are using is narrow enough to reach the adjuster (about 2.5mm or 0.1" diameter) and long enough (at least 12mm or 0.5").
- The trimmer may be at the end of its travel - try turning it several times the other way. It is a 20-turn device.
- The level trimmers do NOT affect the unbalanced outputs.

The Left and Right channels are swapped

- Check that Flip is Off.
- Check that the audio output cables are not reversed.
- Check that the channels are not swapped elsewhere in the system.
- In **Dual AES** mode, ensure that the **AES 1** (or **AES 3**) input is connected to the output on the source equipment for the Left channel data (probably labelled AES 1, AES A or Left) and **AES 2** (or **AES 4**) input is connected to the output on the source equipment for the Right channel data (probably labelled AES 2, AES B or Right). See the manual of the source equipment for information.
- In **Quad AES** or 4-wire DSD modes, ensure that **AES 1, 2, 3 & 4** inputs are correctly connected to the corresponding outputs on the source equipment. See the manual of the destination equipment for information.
- In **SDIF** or **DSD** modes, ensure the **Ch1** and **Ch2** inputs are connected correctly.

One output channel is low or absent

- Check that all cables are connected correctly and not damaged. Damaged cables are a VERY common source of malfunctions!
- Check that the balance is not offset elsewhere in the system.
- If the level trimmers on the rear panel have been adjusted, check that one has not been set much lower than the other.
- In **SDIF** or **DSD** modes, ensure both **Ch1** and **Ch2** inputs as well as the Word Clock are connected correctly.

The output is monophonic

- If the unit is locked to one wire of **Dual AES** or **Quad AES**, the left channel signal will appear on both channels. Select the correct inputs.
- The source may actually be monophonic.
- Check that the signal is not mono'ed elsewhere in the system.

Clicks or crackles occur on the outputs

- Check that all cables are of a suitable type, connected correctly and not damaged. In any multi-wire mode, a broken wire may not prevent the unit locking but will corrupt the data.
- Press the **Coarse Lock** button. If this solves the problem, the source equipment is likely to have a high level of jitter.
- This can be caused by slaving some, but not all, of the system components to a Master Clock. Note that if the unit is being driven by an upsampler, it must be slaved to the upsampler, NOT to a Master Clock.
- If you are using 4-wire DSD, check that the 4 cables are connected correctly.

The Display turns on briefly when a button is pressed, then turns off

- Set the 7-Seg menu item to On.

The unit fails to slave to a Master Clock

- If you are using an AES Reference, connect this to Reference In and set the Ref In menu item to Loop or Loop.t as appropriate. When locked, the unit should display "**r**" followed by the sample rate.
- If you are using Word Clock, connect this to Word Clock In. Set the BNC I menu item to SDIF and the BNC item to RefCl. Press the **BNC** button – the unit should lock and display "**b**" followed by the sample rate.
- Check that the AES Reference In or Word Clock In cable is connected correctly and not damaged.
- Check that the Master Clock is switched on, set to the right sample rate and does not require re-calibration.
- Connect a different piece of digital equipment to test the locking capability of the unit. If the condition persists, contact your Distributor or ***dCS***.

The unit slaves to Word Clock but not AES Reference

- This can be caused by erroneous system messages. Contact your Distributor or ***dCS*** for advice.

The sound has a peculiar tonal balance

- If the sample rate is 48kS/s or lower the De-Emphasis setting may be incorrect. Press the **De-Emphasis** button repeatedly until the mode display shows **A** (**A**uto).
- If this fails to correct the problem, the De-Emphasis flags in the data stream may be incorrectly set (this is rare but it does happen. Press the **De-Emphasis** button to cycle through **5**, **C** and **-** in turn until the correct setting is found. When you change to a different disk or tape, we recommend setting De-Emphasis to **A**uto again.

dCS SUPPORT

I wish ....

If you wish your unit did something it does not, or that this manual told you something it does not, or that we made something we currently do not - tell us. If we can fix it with software, or a manual reprint, and we do so - we will update your unit free of charge. If we do decide to make the thing, we will discuss with you how you would like it to operate.

We value our customers, and we want to make products that do what you want.

If You Need More Help

Contact ***dCS***. Our office hours are 8:00 am to about 7:00 pm, Monday to Friday, UK time (UTC in summer, or UTC + 1hr in winter). 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 15 – ***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, on issues related to ADCs. If you are interested in these, please do not hesitate to contact us.

INDEXES AND SOFTWARE VERSION NUMBERS

This manual is for standard software version 1.5x and P3D unit software v1.36. v1.5x differs from earlier standard software v1.3x and P3D software pre 1.36 in having a more friendly menu structure, with readback on current settings without having to change the settings, and supporting SDIF-3 for DSD.

Definitions of Units

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

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.

dBV Level in decibels, referred to a 1.0V rms sine wave, with no external loading.

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

ADC Analogue to Digital converter, also known as an A/D

DAC Digital to Analogue converter, also known as a D/A

DDC Digital to Digital converter – used for format conversion and some DSP operations that change the data.

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