dCS 900E / 902E

Analogue to Digital Converter

User Manual

Software versions 1.3x to 1.5x

12th June 2000

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

PRODUCT FEATURES

Formats

PCM at 32, 44.1 or 48 kS/s.
dCS 902E only: PCM at 88.2 or 96 kS/s.

Data formats supported are: AES/EBU (XLR), SPDIF (RCA Phono), SPDIF optical (Toslink) and SDIF-2 (BNC).
dCS 902E only: Dual AES (XLR)

Syncing

Master mode or can sync to Word Clock or AES reference.
Sync to video option available

Functions

Very high performance ADC, free from gain ranging
Multichannel Sync capability
Noise shaping truncation (1st, 3rd, 9th order)

Test Generator

High quality (160 dB) tone generator. Can be noise shaped truncated.
dCS 902E only: Variable tone frequency and level.

Ease of Use

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

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

CONTENTS

About this Manual

Note that there is a fuller Contents at the end of the manual (page 63), 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 is designed to be helpful. If there are points you feel we could cover better, or that we have missed out - please tell us.

Warranty

Your ***dCS 900*** or ***dCS 902*** 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.

USING YOUR *dCS 900 / 902* FOR THE FIRST TIME

Product Overview

The ***dCS 900*** and ***902*** ADCs (Analogue to Digital Converters) are high performance converters designed for studio and live recording applications. They are designed to produce very high standard digital output that may be used directly or archived. AES3, SPDIF and SDIF-2 PCM formats are all supported. Multiple units may be slaved together 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 “panel lockout” feature.

The unit is highly software based, and more functions and features are 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 900 or ***dCS 902***
User Manual
Function Menu Guide
Mains Lead
2 Spare Fuses
Remote cable
Remote software

Mains Voltages

The ***dCS 900 / 902*** 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***.

Installing Unit in a Rack

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

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

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

[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.

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 unit will test itself for a few seconds while displaying:

tESt

Next, the main display will briefly show:

- - - - -

or

- - - - 2

and then the sample rate, for example:

44.1

do this: Connect a signal source to the analogue inputs.

do this: Connect an output (e.g. from **AES1**) to your system or a DAC.

do this: Press the **Sample Rate** button (left hand end button) to get the sample rate you want.

do this: ***dCS 902*** only: Press the **Output Format** button (right hand end) to get the format you want.

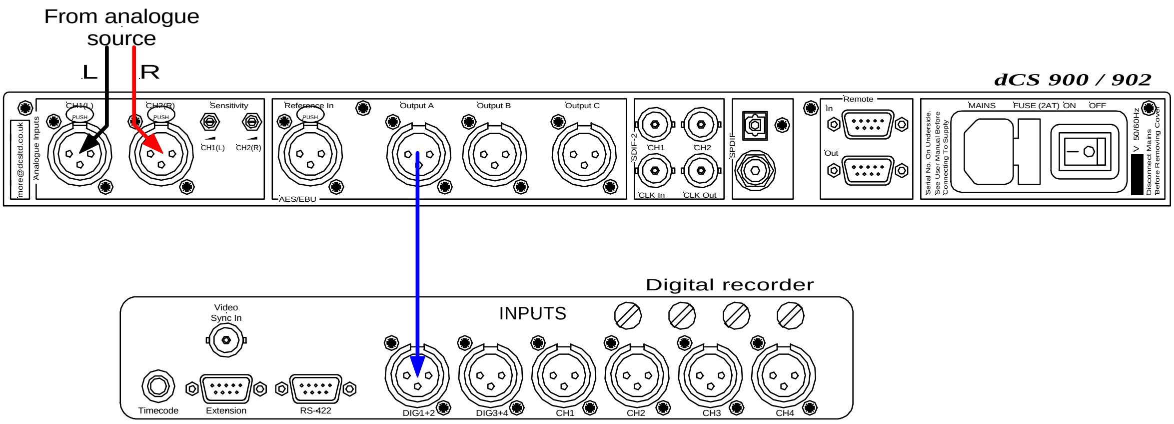

Figure 1 – Recording with a ***dCS 900 / 902***

Set up like this, the ***dCS 900 / 902*** will operate in Master mode, and the system it is connected to will (have to) lock to it. You should have audio.

Use any output at 32 kS/s or 44.1 kS/s or 48 kS/s.

dCS 902 only: Use any output for double speed data at 88.2 kS/s or 96 kS/s.
Use (**AES1** + **AES2**) for Dual AES at 88.2 kS/s or 96 kS/s

If you want to record less than 24 bit data, press the **Word Length** button repeatedly until the required Word Length is displayed and press the **Noise Shaping** button repeatedly until Auto is displayed.

Note that all the outputs are active simultaneously on the ***dCS 900 / 902***. If the mode the unit is in needs them to be different, they will be – otherwise they will be the same, and may all 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

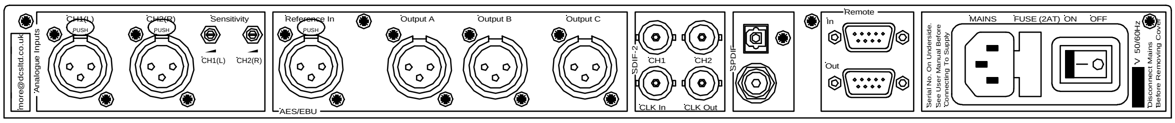

Figure 2 – Rear Panel

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

Balanced Analogue Inputs **3 pin XLR female (2 off)**

Input Level Adjustment **(trimmers)**

Two 20-turn potentiometers set the full scale input levels. These are factory preset for full scale with input levels of +20dBu. If necessary, adjust with a suitable trim tool or a small screwdriver. Turn clockwise for increased gain. Take care to ensure the stereo inputs remain in balance. The trim range is ±6dB.

Reference In **3 pin XLR female**

AES C (Reference) Digital Output **3 pin XLR male**

Reference In is an AES/EBU reference input for synchronising the unit to a Master Clock. **AES C** is a Reference loop through, directly coupled to it. A terminating resistor may be turned on or off, using the menu, if several units are to be daisy chained.

AES A & B Digital Outputs **3 pin XLR male (2 off)**

Two AES/EBU outputs which may be used independently.

dCS 902 only: They may also be used as a **Dual AES** pair at 88.2 or 96kS/s.

SDIF-2 CH1, CH2 Data **BNC (2 off)**

These outputs are for **SDIF-2** encoded PCM. They are both TTL level signals for a 75 ohm line.

SDIF-2 Clk In **BNC**

SDIF-2 Clk Out **BNC**

This pair take in and give out Word Clock. **Clock In** is terminated and **Clock Out** is regenerated internally, so these lines can be used for daisy chaining many units together.

SPDIF Outputs **RCA Phono & Toslink optical**

The RCA Phono connector should be used with a 75 ohm line.

The Toslink connector should be used with a Toslink fibre-optic cable designed for digital audio use. Pull out the plastic cover before use.

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 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. In addition, the unit may be software upgraded without removing the lid by downloading new software via the **Remote In** port.

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

The underside of the unit will have a label on that contains a number such as 900 1A2 3B4 5C6 7D8 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.

Digital Data Formats

The unit provides five digital data i/o formats,

AES/EBU	(often referred to as AES3)
Dual AES	(part of the AES3 spec)
SDIF-2	
SPDIF (electrical)	
SPDIF (optical)	

For all formats, the incoming Channel Status and User messages are handled according to a priority system – they are either passed through, where this is sensible, or generated and inserted by the unit.

The enhanced AES/EBU interface is fully implemented. Each channel has its own parity and data validity bit, as well as User and Channel Status messages. Cyclic Redundancy Counts (CRC's) are generated from the Channel Status message. The Dual AES interface allows a 96 or 88.2 kS/s 24 bit signal to be coded as two standard 48 or 44.1 kS/s 24 bit AES data streams, recorded as four channels on a recorder with standard capacity, replayed and decoded back into a single data stream / channel pair.

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

The SPDIF interface has no CRC's – as per definition. The data structure for electrical and optical interfaces is identical.

Front Panel

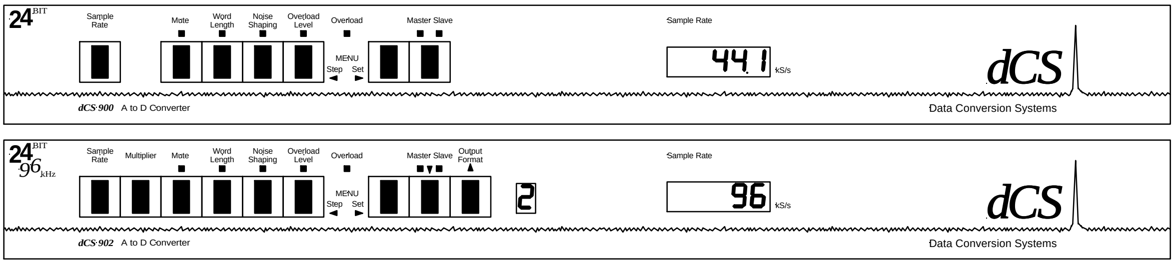

Figure 3 – Front Panel

The ***dCS 900 / 902*** uses a combination of front panel buttons for frequently changed functions and a step through menu for features you might set and forget.

Sample Rate

Multiplier

Press the **Sample Rate** button repeatedly to cycle through the sample rates. ***dCS 900***:

48 ... 44.1 ... 32 ... 48 ... etc.

dCS 902:

96 ... 88.2 ... 48 ... 44.1 ... 32 ... 96 ... etc.

The ***dCS 902*** also features a **Multiplier** button which allows you to step through more quickly. Press the **Multiplier** button repeatedly to cycle through in one of the following sequences, depending on the starting sample rate:

48 ... 96 ... 32 ... 48 ... 96 ... etc.

44.1 ... 88.2 ... 44.1 ... 88.2 ... etc.

To change sample rates quickly, use the two buttons together. For example, to change from 88.2 kS/s to 96 kS/s press **Sample Rate** once then **Multiplier** once. Do not press the buttons too fast as a delay is built in to the software. The sample rate selected is shown on the LED display in the centre of the panel.

Mute

This button forces a mute, in addition to the automatic ones. The digital outputs are automatically muted at power up and when the sample rate is changed or the unit is locking to a reference source. A forced mute is indicated by the mute LED (above the mute switch) lighting up. In normal use, pressing the **Mute** button mutes the digital outputs and lights the mute LED. Pressing the **Mute** button again unmutes the ADC, as long as no automatic mute is being applied.

Word Length

Noise Shaping

The AES/EBU format accommodates data up to 24 bits. If a shorter word is needed and the extra bits are just ignored, the result is typical "digital" sound due to the abrupt chopping off of the low level signal information.

To avoid this, the ***dCS 900 / 902*** allows proper truncation of the data and uses Noise Shaping to maintain low level performance. See the section on "**Word Length Reduction"** on page 50 for some background on this. If you do use **Word Length** truncation, make sure that **Noise Shaping** is not set to **OFF** without realising it.

Pressing the **Word Length** button repeatedly cycles the word length through the sequence:

24, 23, 22, 21, 20, 19, 18, 17, 16, 24, etc.

The Word Length is briefly shown on the main display, and if a setting other than the maximum is set, the word length LED (above the button) lights.

Noise Shaping is a technique which improves the noise performance of the ADC in the audio band by moving the quantisation noise energy (introduced by reducing the word length) from the middle of the band, where the ear is most sensitive, to the top end or ultrasonic region, where the ear is less sensitive or insensitive. See the section **"Word Length Reduction"** on page 50 for more background.

Pressing the **Noise Shaping** button repeatedly cycles the unit through 5 noise Shaping characteristics. The characteristic is shown briefly on the main display.

Auto	Unit sets noise shaping automatically, depending on word length: 24 bits – no noise shaping 20 to 23 bits – 1^{st} order noise shaping 16 to 19 bits – 3^{rd} order noise shaping
Off	No noise shaping
1st	1^{st} order noise shaping
3rd	3^{rd} order noise shaping
9th	9^{th} order noise shaping

The noise shaping LED (above the button) lights when the setting is other than Auto.

Overload Level

Menu Step

The **Overload Level** button is dual function – on its own (**blue** type on the front panel) it sets the level at which overloads are detected by the unit. With the other menu buttons (**white** type on the front panel) it is the menu **Step** button.

Overload detection is normally set to full scale. The detection level may be reduced in 0.1dB steps down to -3dB0 by pressing the **Overload Level** button repeatedly or holding it down. The set level is shown on the display for a few seconds. The overload level LED (above the button) lights when the setting is other than full scale (0.0dB0).

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

Overload Indicator **(Overload LED)**

This overload LED lights for a few seconds when the set overload level is exceeded by a signal peak. The detection circuitry monitors both input and digital filtering circuitry for overload conditions. The analogue input sensitivity trims mounted on the rear panel should be set so that the overload indicator does not light on signal peaks.

The overload indication given by the ***dCS 900 / 902*** is comprehensive. The detection circuitry monitoring the digital filter does not simply check the final output word but all the data from which the output word is formed. If any of these overload (this may not be apparent from the output data), an overload is flagged.

The filter itself has sufficient numerical accuracy that if the input data is not overloaded, the filter computations cannot generate an overload - only a raw data overload can cause an error. The overload indication is thus much more accurate than any external meter based indication - for this reason it is stored in the AES/EBU validity bit for later reference.

Menu Set

The **Set** button is used with the other menu buttons (**white** type on the front panel). See the section **"The Software – The Menu"** on page 18.

Master/Slave **Menu Down**

The **Master/Slave** button is dual function – on its own (**blue** type on the front panel) it sets the clocking mode (master or slave). With the other menu buttons (**white** type on the front panel) it is the menu **Down** button.

In Master mode, the calibrated voltage controlled crystal oscillators (VCXOs) inside the unit generate an accurate sample rate. The LED labelled Master will be lit to indicate this. If a Master Clock is available, this may be connected to the **Reference In** connector (for AES/EBU reference) or the **Clk In** connector (for SDIF-2 Word Clock). To slave the unit to the Master Clock, press the **Master/Slave** button. The unit will attempt to lock to the Reference - this will take a few seconds. If lock is achieved, the Slave LED will light up brightly and the Master LED will turn off. To return to Master mode, press the **Master/Slave** button again.

If both AES Reference and Word Clock are connected, pressing the **Master/Slave** button cycles through the sequence:

Master ... AES Reference ... Word Clock ... Master ... etc.

If the active reference source is lost, the unit will select the next option in the sequence.

If the Auto-Slave option (see page 21) is turned On the unit will automatically slave when a suitable reference is connected. If both AES Reference and Word Clock are connected, AES Reference takes priority. Word Clock may be selected by pressing the **Master/Slave** button – it moves down the priority list.

dCS 902 only: Once slaved, the unit can internally multiply the reference input sample rate by 2, if required, by pressing the **Multiplier** button. The Master Clock must be set to a suitable sample rate:

Master Clock Sample Rate (kS/s)	*dCS 902* Sample Rate (kS/s)
32	32
44.1	44.1 or 88.2
48	48 or 96
88.2	88.2
96	96

Table 1 - Reference Clock and Sample Rates

For Menu operation as the **Down** button, see the section**"** on page 18.

Output Format **Menu Up**

dCS 902 only: The **Output Format** button is dual function – on its own (**blue** type on the front panel) it sets the output format to Single AES (1) or Dual AES (2). With the other menu buttons (**white** type on the front panel) it is the menu **Up** button.

Pressing the **Output Format** button repeatedly causes the output format to cycle through the allowed options from the sequence:

Single AES ... Dual AES ... Single AES ... etc.

For sample rates of 48kS/s or less, Single AES is the only option.

IMPORTANT! *If the* **Output Format** *is set to* 2 *(Dual AES), do not use the* **SPDIF** *or* **SDIF-2** *outputs.*

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

Mode Display

dCS 902 only: The single digit LED mode display to the right of the **Output Format** button shows the output format:

Display	Output Format
1	Single AES
2	Dual AES

Table 2 - Output Data format indication, higher sample rates

When the Output Format is selected, the main display briefly shows the format code:

A1	for Single AES, Standard speed encoding
b1	for Single AES, Double speed encoding
b2	for Dual AES, Standard speed encoding

In Single AES mode, the same data stream is available on both AES outputs.

In Dual AES mode, the data stream is available on **AES A** & **AES B** outputs.

dCS equipment encodes messaging into the various data streams to enable receiving equipment to tell what is going on, and to decide which wire is which, in the unlikely event of user wiring errors. Not all equipment from other manufacturers does this, so:

IMPORTANT! *Take care when connecting Dual AES as it is easy to connect the wires in the wrong order. If this is not detected, the Left and Right channels will be swapped on the recording. Numbering each connector is a sensible precaution.*

Sample Rate Display

The main LED display generally shows the sample rate, in kS/s. When other parameters are set, it briefly shows the new setting (Word Length, Noise Shaping, 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, the display also indicates which input connector it is slaved to.

xxx	The sample rate, in kS/s (32, 44.1, 48, 88.2 or 96).
b xxx	Slaved to **Clk 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 **Clk Out** to **Clk in**.

Important error messages are given below – a full list is given in the section **Error Codes and Messages** on page 58.

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 58 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 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 900 / 902*** 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**. 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 for:

Step	otherwise **Overload Level**
Set	
Down	otherwise **Master/Slave**
Up	otherwise **Output Format** (***dCS 902*** only)

Entering the Menu

The Menu is entered by holding down the **Step** button 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 change its setting. This either toggles the previous 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 4 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.
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.
dCS 900E : Use **Master/Slave** button to alter **RS232** address.
dCS 902E : Use **Menu Up** & **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.

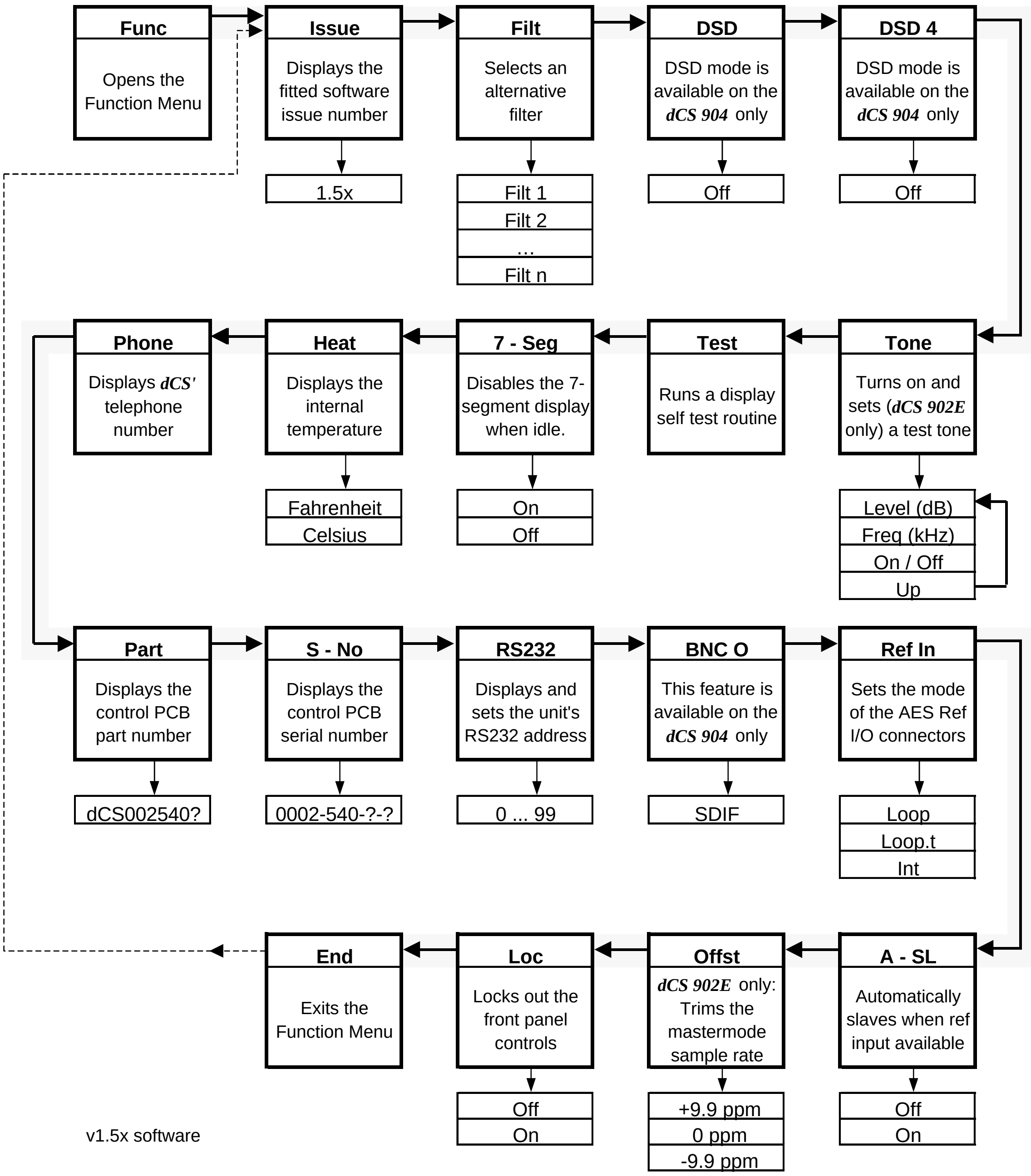

Figure 4 – Menu Sequence

Menu Items

Issue

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

Filt

Selects one of several anti-alias 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, degrading the alias performance but sharpening the impulse response. This affects the stereo or multi-channel image. Different filters may be appropriate for different material.

DSD

DSD 4

DSD and DSD 4 modes are only available on the ***dCS 904***. In the ***dCS 900*** & ***902***, they are permanently set to Off.

Tone

This accesses a sinewave test generator. On the ***dCS 900***, the tone is fixed at 1kHz and -18dB0. Press **Set** to toggle it On and Off.

On the ***dCS 902***, the tone level and frequency can be adjusted. Pressing **Set** enters a submenu, which accesses the following functions:

Level	The tone level, from –120dB0 to 0dB0. It can be changed in 0.1dB steps using the **Up** and **Down** buttons.
Freq	The tone frequency, from 0.001kHz (1Hz) to just under half the sample rate. It can be changed using the **Up** and **Down** buttons. The step size is 1Hz for tones up to 100Hz, 10Hz up to 1kHz and 100Hz above 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.

At power up, the tone generator is set to –18dB0, 1kHz and turned Off.

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 58 for more specific information.

7-Seg

Disables the 7 segment LED display. When set to Off, the display turns off 4 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.

Heat

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

Phone

dCS telephone 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 to tell us this number if you call us for help.

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. On ***dCS 900***, press (or hold down) the **Master/Slave** button to change this setting.

IMPORTANT! *Each unit in a daisy chain MUST be set to a different RS-232 address.*

BNC O

This feature is available on the ***dCS 904*** only. In the ***dCS 900*** & ***902***, it is permanently set to SDIF.

Ref In

Sets the mode of the **AES Reference In** / **AES C** Out connectors. The options are:

Loop	Loops the **Reference In** through to the **AES C** output, with no termination resistor (termination is then about 1kohm, so several units can be daisy chained).
Loop.t	As above, but terminates the input. Use at the end of a daisy chain.
Int	The output (and input in parallel – beware!) is internally driven, with the same signal as **AES 1**.

A-SL

Turns Auto-slaving On or Off. When set to On, connecting an AES/EBU reference or a word clock in causes the unit to slave and lights the Slave LED. If both are present, the unit picks the highest priority one (AES/EBU) unless the **Master/Slave** button is used to move down the priority list. When set to Off, the unit does not react when a reference is connected.

IMPORTANT! *Ensure the* Ref In *menu item is set to* Loop *or* Loop.t *before connecting an AES Reference.*

Offst

dCS 902 only: Trims the appropriate VCXO frequency in master mode, by up to ±9.9ppm in 0.1ppm steps. Use the **Up** and **Down** buttons to change this setting, hold a button down to accelerate the change. The trim is remembered.

Loc

Panel Lock , normally Off. Set to On to prevent unauthorised changes using buttons. The menu has to be accessed in the usual way to turn the lock off again.

End

Exits the menu.

TYPICAL APPLICATIONS

Six Channel Set Up Sync'ed to AES Reference

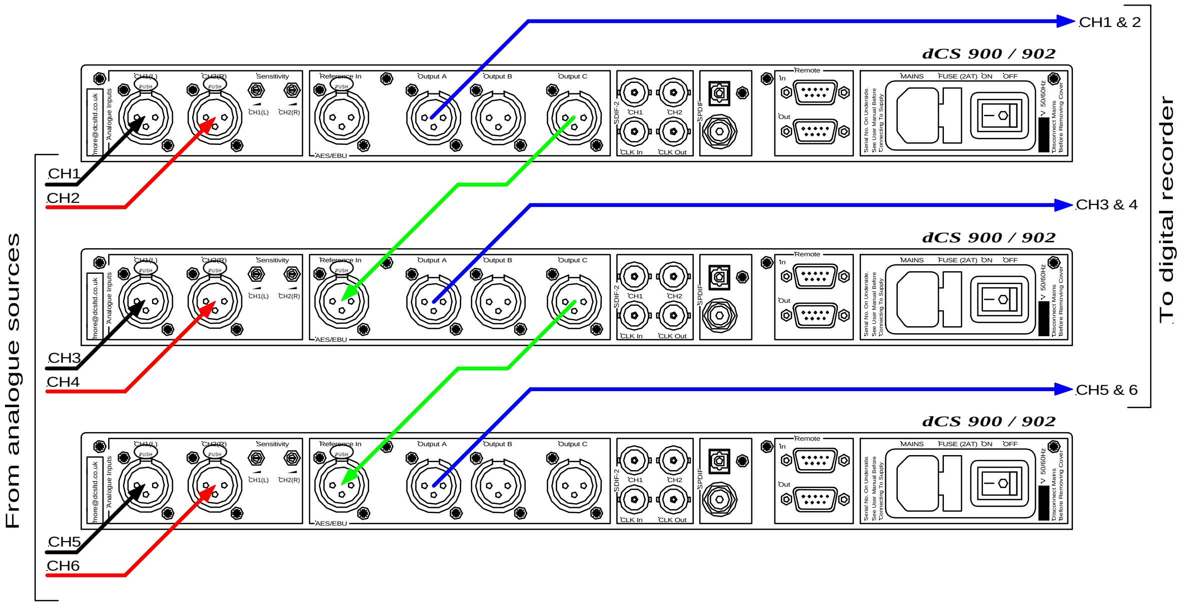

Figure 5 – Six Channel Set Up Sync'ed to AES Reference

do this: For the top ADC, set to the sample rate you wish to record, set Ref In to Int (Internal sync) and select a filter.

do this: For the middle ADC, set Ref In to Loop, turn A-SL (Autoslave) On and select a filter.

do this: For the bottom ADC, set Ref In to Loop.t, turn A-SL On and select a filter.

do this: For all ADC's, if you are recording less than 24 bit words, set **Word Length** as required and set **Noise Shaping** to **Auto**.

The units self align quite accurately (see the section **"Sample Alignment"** on page 32 onwards).

This arrangement will work with one, two or more ADC's (the top one must be set to Int).

You can record from the SDIF-2 output or one of the SPDIF outputs if preferred.

Six Channel Set Up Sync'ed to Master Clock (AES)

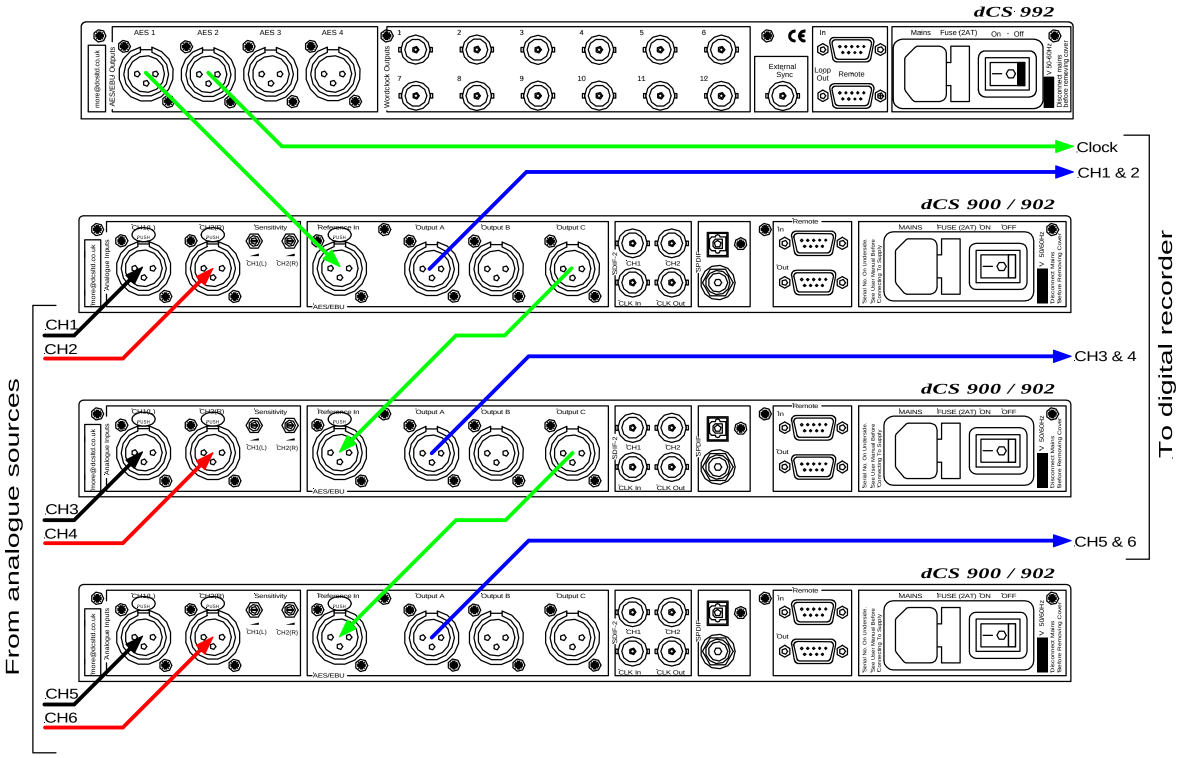

Figure 6 – Six Channel Set Up Sync'ed to Master Clock (AES)

do this: Set the ***dCS 992*** Master Clock to the sample rate you wish to record at and set the recorder to slave to it.

do this: For all 3 ADC's, set A-SL (Autoslave) to On and select a filter. If you are recording less than 24 bit words, set **Word Length** as required and set **Noise Shaping** to **Auto**.

do this: For the ADC's processing CH1 – 4, set Ref In to Loop. For the bottom ADC, set Ref In to Loop.t.

The Master Clock helps reduce jitter. The units self align quite accurately (see section **"Sample Alignment"** on page 32 onwards). This arrangement will work with one, two or more ADC's.

You can record from the SDIF-2 output or one of the SPDIF outputs if preferred.

Six Channel Set Up Sync'ed to Word Clock

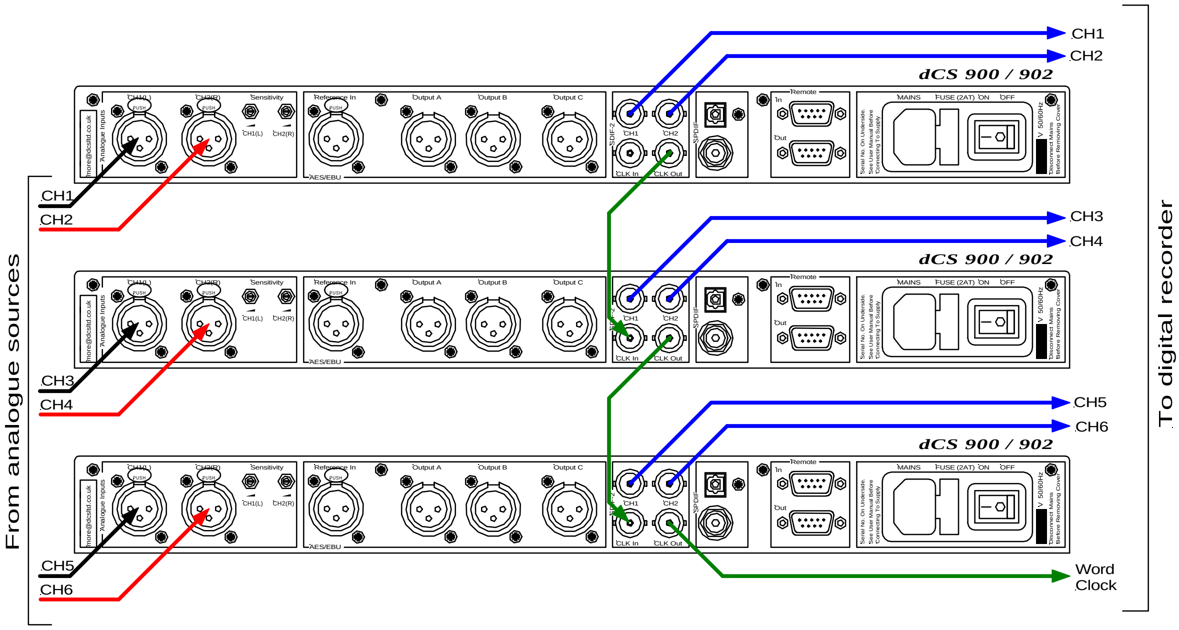

Figure 7 – Six Channel Set Up Sync'ed to Word Clock

do this:	For the top ADC, set to the sample rate you wish to record, set Ref In to Int (Internal sync) and select a filter.
do this:	For the other ADC's, turn A-SL (Autoslave) On and select a filter.
do this:	If you are recording less than 24 bit words, set **Word Length** as required and set **Noise Shaping** to **Auto** on all ADC's.

The data streams will be synchronous.

This arrangement will work with one, two or more ADC's (the top one must be set to Int).

You can record from the AES outputs or one of the SPDIF outputs if preferred.

Six Channel Set Up Sync'ed to Master Clock (Word Clock)

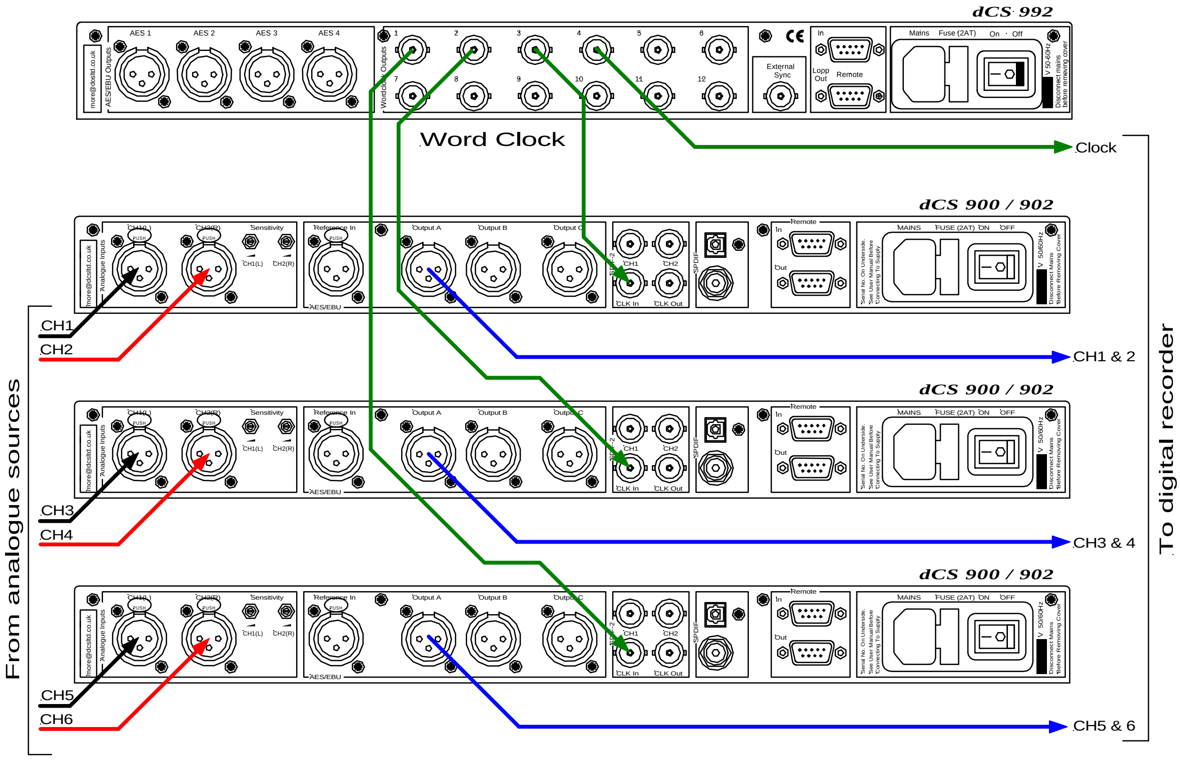

Figure 8 - Six Channel Set Up Sync'ed to Master Clock (Word Clock)

do this: Set the ***dCS 992*** Master Clock to the sample rate you wish to record at and set the recorder to slave to it.

do this: For all 3 ADC's, set A-SL (Autoslave) to On and select a filter. If you are recording less than 24 bit words, set **Word Length** as required and set **Noise Shaping** to **Auto**.

The Master Clock helps reduce jitter. This arrangement will work with one, two or more ADC's.

You can record from the SDIF-2 output or one of the SPDIF outputs if preferred.

Operating Several Units on One Remote Chain

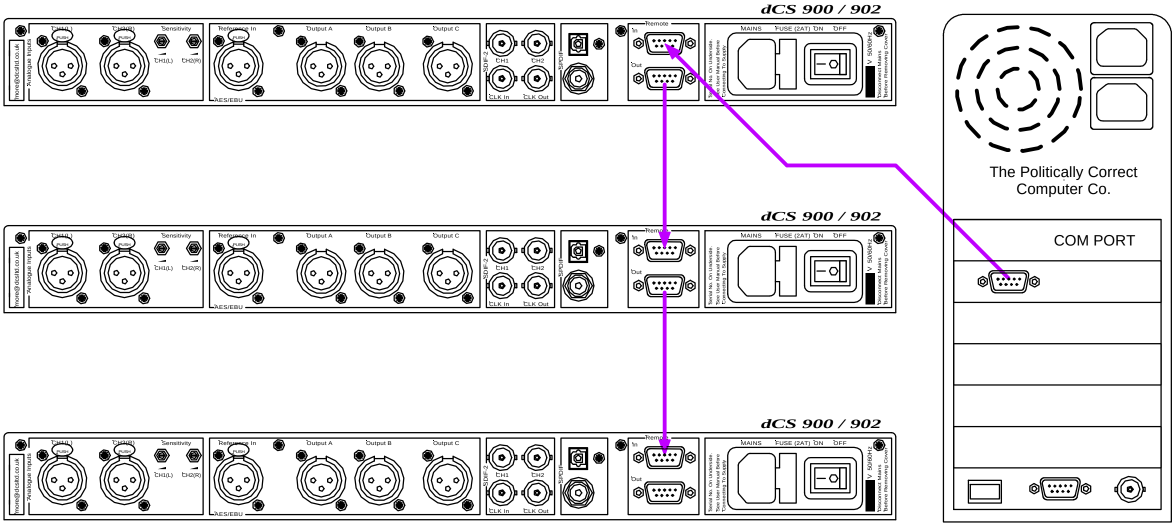

Figure 9 – Multi-unit Remote Daisy Chain

The PC can control several ***dCS*** units (up to about 5) of different types 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.

See **"Remote In & Out"** on page 11 for cable details.

Recording Dual AES on a 4-Track Recorder

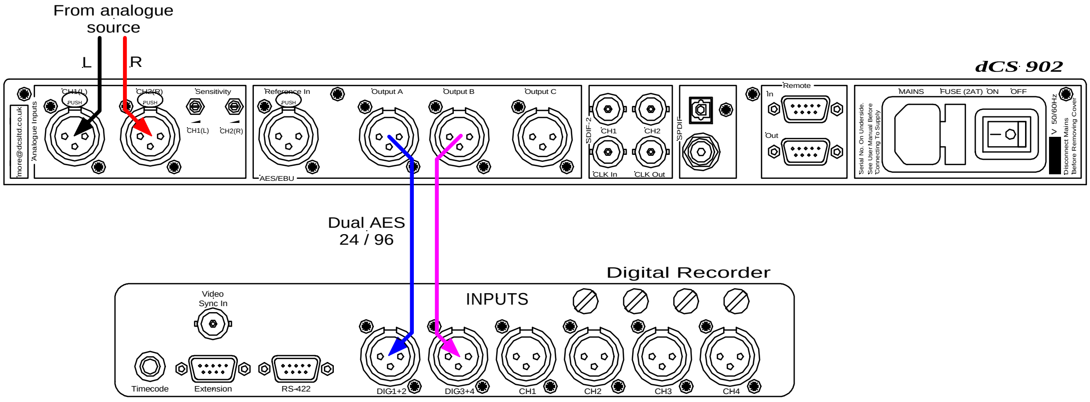

Figure 10 – Recording Dual AES on a 4-Track Recorder

do this: Set the ***dCS 902*** to the sample rate you wish to record at (88.2 or 96kS/s) and set the **Output Format** to **2**. Set Ref In to Int and select a Filter. If you are recording less than 24 bit words, set **Word Length** as required and set **Noise Shaping** to **Auto**.

do this: Replay the tape through a Dual AES capable DAC such as ***dCS 952***.

dCS 900 / 902 TECHNICAL INFORMATION

Anti Alias Filtering

The ***dCS 900*** & ***902*** offer a choice of 4 anti-alias filters on most sample rates. These filters affect the ultrasonic part of the spectrum - 20 kHz upwards.

The unit is an ADC, with an output data rate set by the interface standard used. The bandwidth of the input stages and oversampling converter used is high, and so any signals that that are in the input signal, up to a MHz or so, will be aliased[3] back into the output signal if they are not removed by filtering. The demands on this anti-alias 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 prevent aliasing about 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 to alias irrevocably into the output data. Once a signal has aliased, it cannot be corrected. 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 some degree of aliasing 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 aliasing, but the worst energy smear. Then as the number increases the smear decreases, but the aliasing 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.

The ***dCS 900*** & ***902*** use 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[4], depending on the filter being used. The stop band is typically below –110 dB and can be as low as –130 dB.

[3] See, for example "Principles of Digital Audio", 3rd Edition, by Ken C Pohlmann (McGraw-Hill Inc, 1995)
[4] 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 output performance of an ADC.

The highest quality clocks that are available are crystals, so we use these. In Master mode, the ***dCS 900*** & ***902*** use one of two on-board voltage controlled crystal oscillators (VCXOs) as clock sources – one for 48 kS/s related outputs and one for 44.1 kS/s related outputs. When an external clock is applied for Slave operation, the internal VCXO is 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 slave clocks are used. A consequence of the narrow bandwidth is that it takes quite a long time for the PLL to lock to a new clock frequency – of the order of 2 seconds. The PLL uses DSP assistance to keep this time acceptable.

Internal clock

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

dCS 902 only: The VCXO frequency can be trimmed by using the Offst function in the menu (see page 22) – each VCXO is independently adjustable

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.

Sample Alignment

The ***dCS 900*** & ***902*** align samples such that Word Clock out aligns with AES3 samples out, the rising edge of word clock 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. The scope shots below were taken in Master mode.

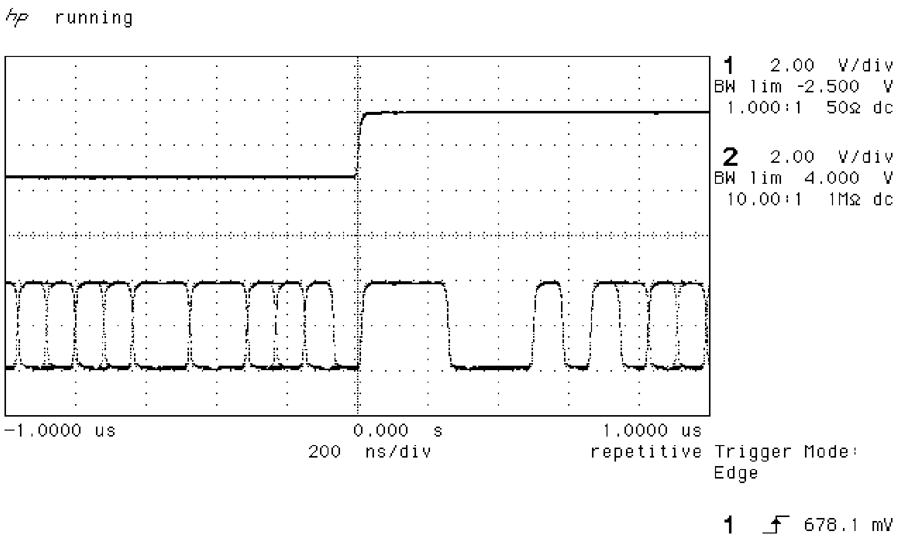

Figure 11 – Word Clock and AES3 outputs, 96 kS/s

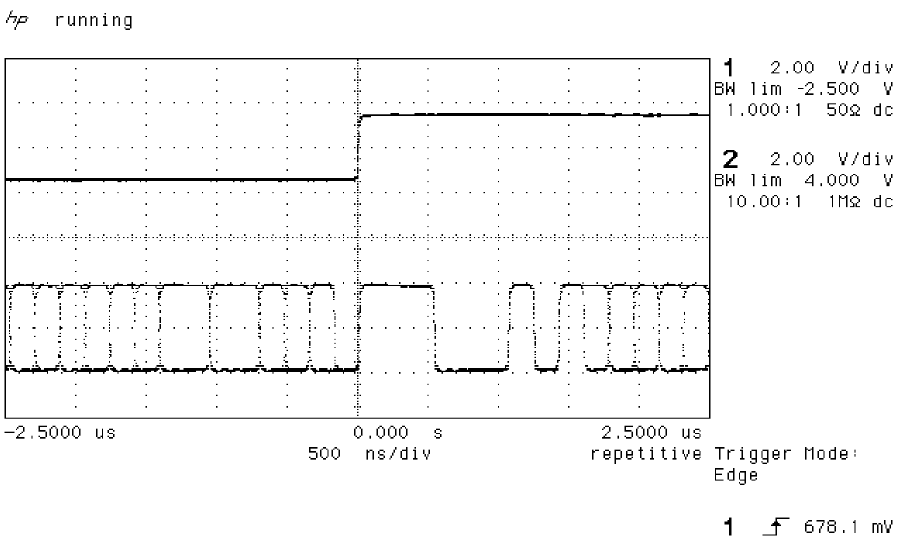

Figure 12 – Word Clock and AES3 outputs, 44.1 kS/s

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 40 nsecs. The scope shots below were taken with the unit sync'd to Word Clock in.

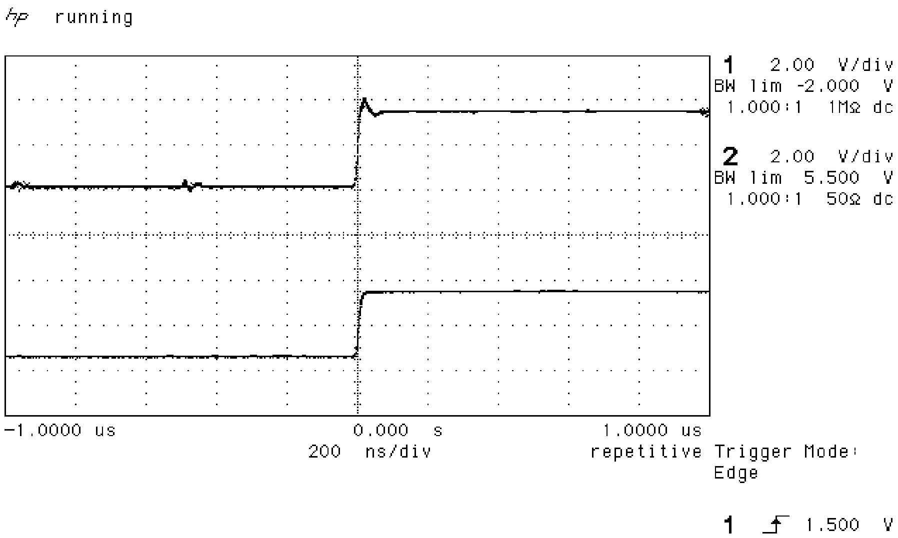

Figure 13 – Word Clock in to Word Clock out, 96 kS/s

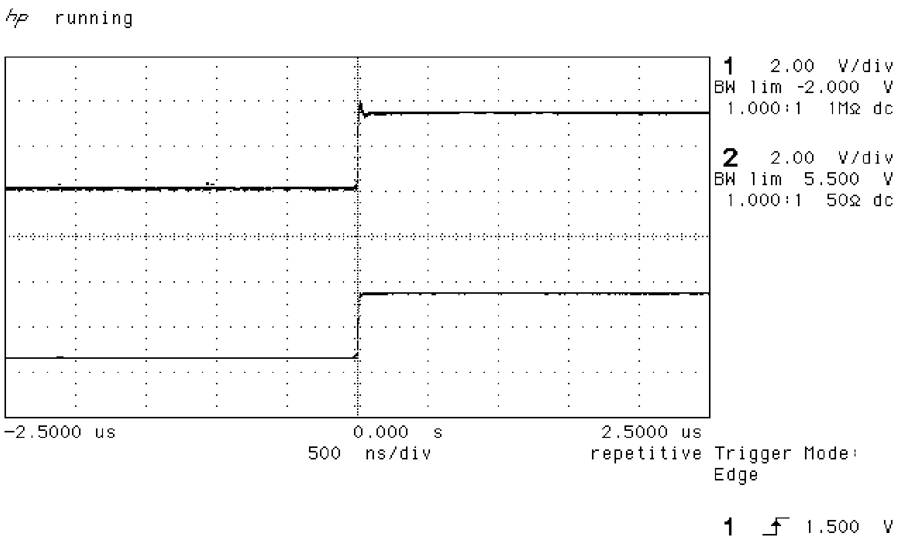

Figure 14 – Word Clock in to Word Clock out, 44.1 kS/s

AES3 in and out are related as below, where they are at the same sample rate, and the AES3 input is used as a sync source. The alignment is better than 40 nsecs. Input is at the top of the displays, output is at the bottom. Signals are at the sockets on the ***dCS 900*** & ***902***, and the unit was slaved to AES Ref In.

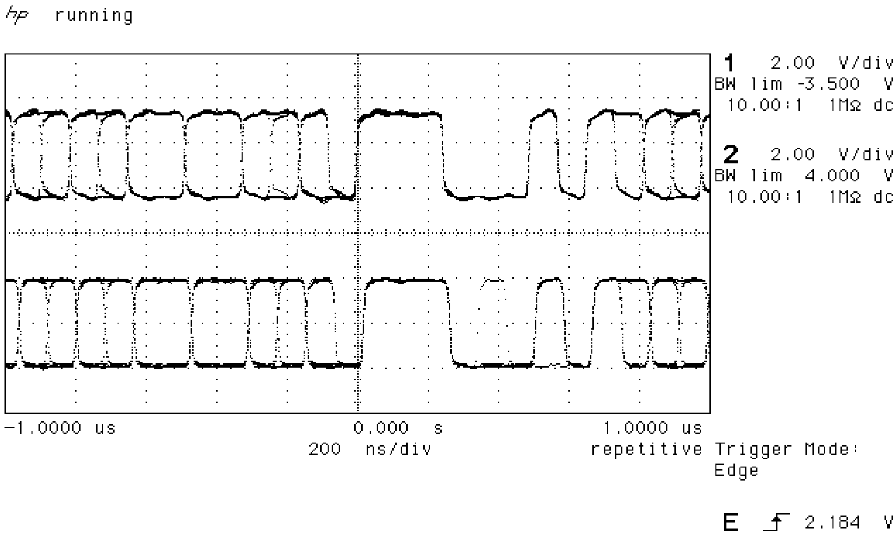

Figure 15 – AES3 in to AES3 out, 96 kS/s

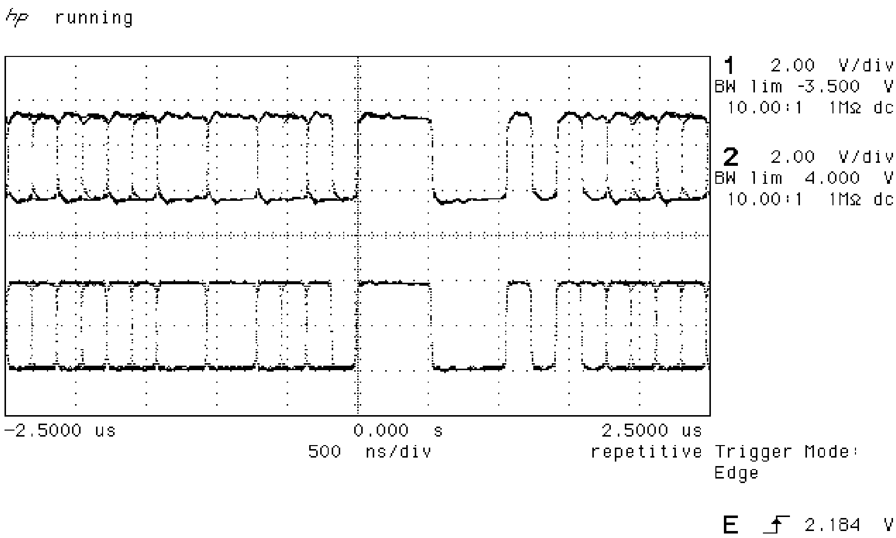

Figure 16 – AES3 in to AES3 out, 44.1 kS/s

AES3 data out is also related to the phase of Word Clock in. The scope shots below were taken with the unit sync'd to Word Clock.

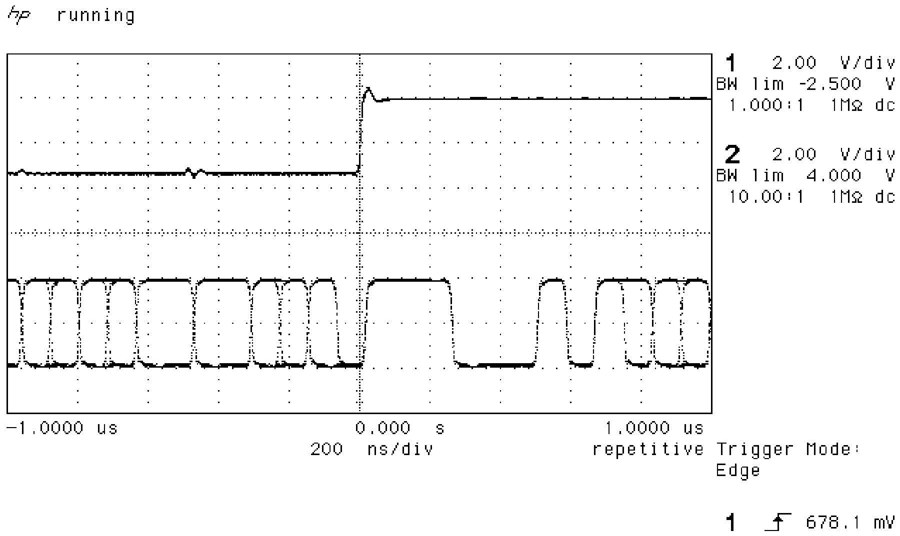

Figure 17 – Word Clock in to AES3 out, 96 kS/s

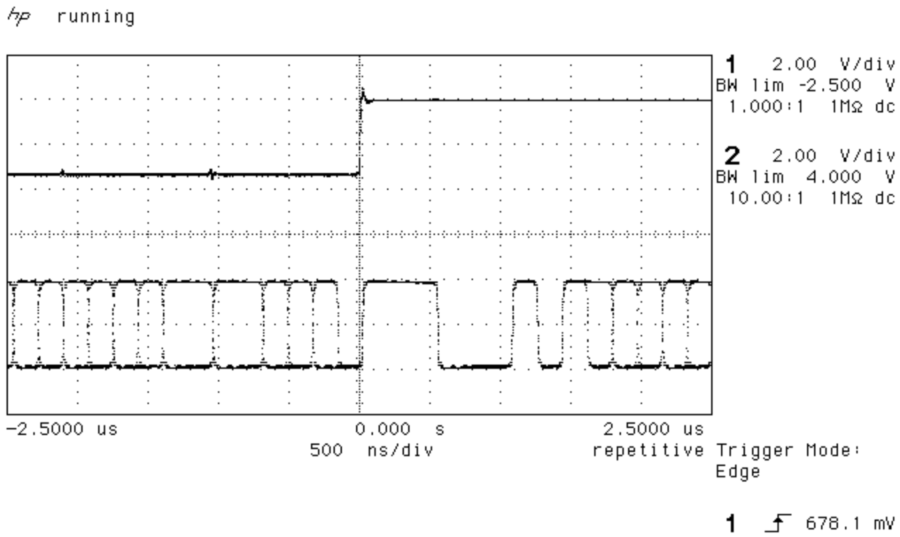

Figure 18 – Word Clock in to AES3 out, 44.1 kS/s

Noise Shaping

The ***dCS 900*** & ***902*** use noise shaping[5] that is optimised to the F weighting curve[6]. It does not affect signal frequency or transient response, but shapes the frequency response of errors (Q noise, or truncation errors) so that they fall as much as possible in the less sensitive part of the spectrum. For all the major sample rates (32 kS/s, 44.1 kS/s, 48 kS/s, 88.2 kS/s, 96 kS/s) the noise shapers have been individually optimised. The 1st, 3rd, and 9th shapes for 44.1 kS/s agree well with Wannamaker's published results[7].

Noise Shaping adds more noise power, but because of the shaping it is perceived as lower noise. There is a compromise to be drawn – as more aggressive shaping is used, more noise is added, and less perceived improvement occurs. In practice, things stop improving much above the 9th order. The increased real noise power can cause (small) clicks in editing, if this is carried out after the shaping. For this reason, noise shaping should be used as late as possible in the mastering process – we recommend recording at the very highest possible sample rate and resolution, and only reducing either at the latest possible minute.

If, however, you have to reduce word length, the perceived noise gain (taking into account the ear's response) and the actual increase in noise (mainly out of band) in given in the table below.

Sample Rate (kS/s)	Perceived Gain, F weighted, 1st Order (dB)	Actual Increase in Noise, 1st Order (dB)	Perceived Gain, F weighted, 3rd Order (dB)	Actual Increase in Noise, 3rd Order (dB)	Perceived Gain, F weighted, 9th Order (dB)	Actual Increase in Noise, 9th Order (dB)
32	-3.3	1.9	-7.5	4.2	-8.1	6.1
44.1	-5.5	2.4	-10.5	6.9	-17.9	23.4
48	-6.2	2.5	-11.7	7.6	-21.0	23.8
88.2	-11.1	2.8	-23.8	11.3	-42.2	24.0
96	-11.8	2.9	-25.7	11.3	-45.3	22.5
176.4	-17.0	3.0	-40.6	12.6	-63.0	21.8
192	-17.7	3.0	-42.8	12.6	-65.9	21.8

Table 3 – Noise Shaper Gain by Order and Sample Rate

The 3rd order shaping tends to follow the E weighting curve, by chance. The 9th order is very aggressive, and can give very large gains at the higher sample rates. For example, 176.4kS/s or 192kS/s material truncated to 16 bits (so it can be stored on a DA-88 or ADAT) loses nothing in the audio band in terms of perceived noise, with 3rd or 9th order shaping. For more information on this topic, either see section **"Word Length Reduction"** on page 50 or read the references below.

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

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

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

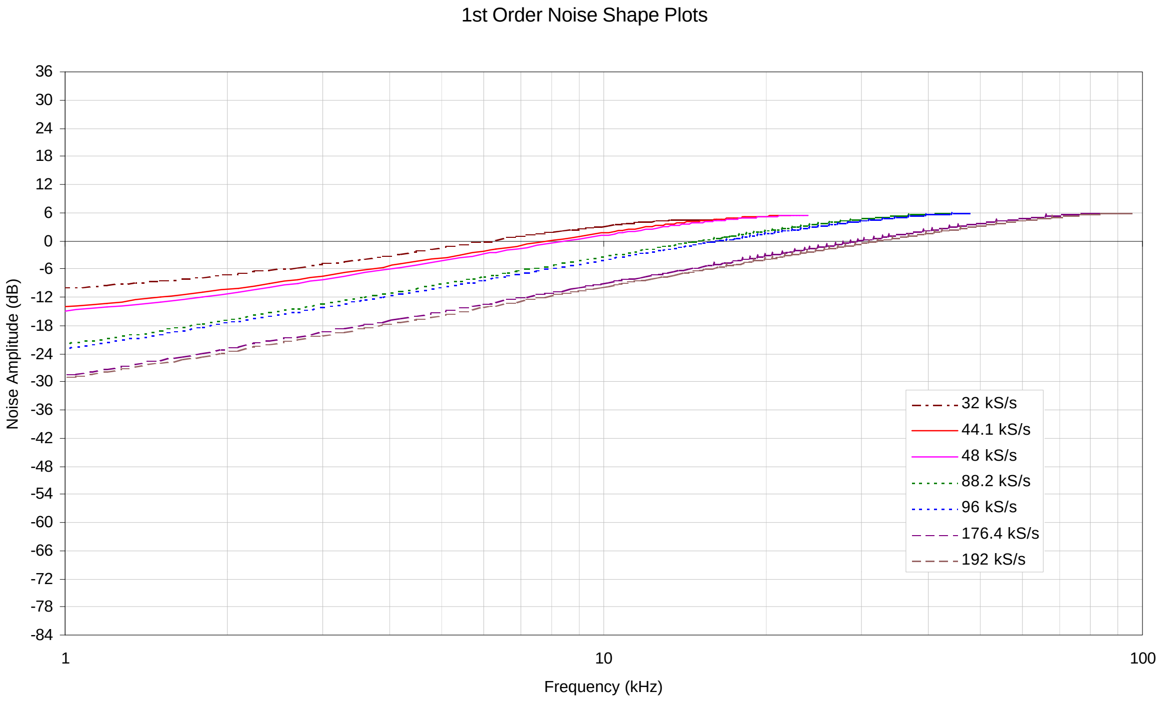

Figure 19 – 1st Order Noise Shapers implemented on ***dCS 90X***

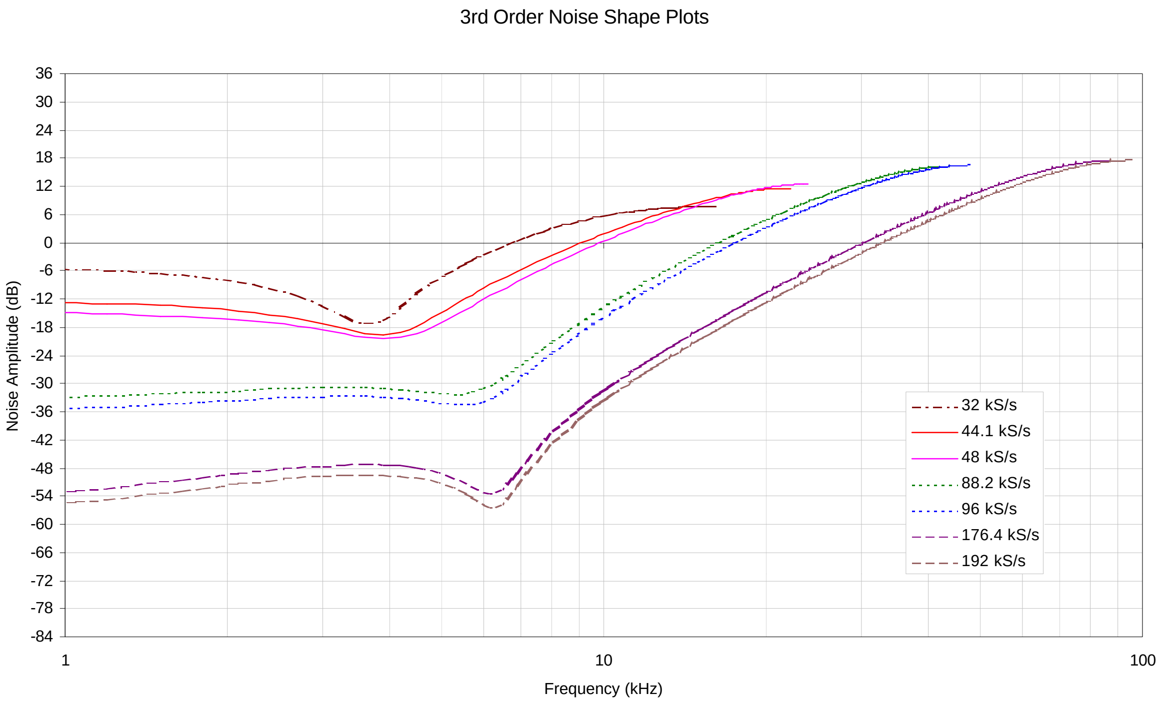

Figure 20 – 3rd Order Noise Shapers implemented on ***dCS 90X***

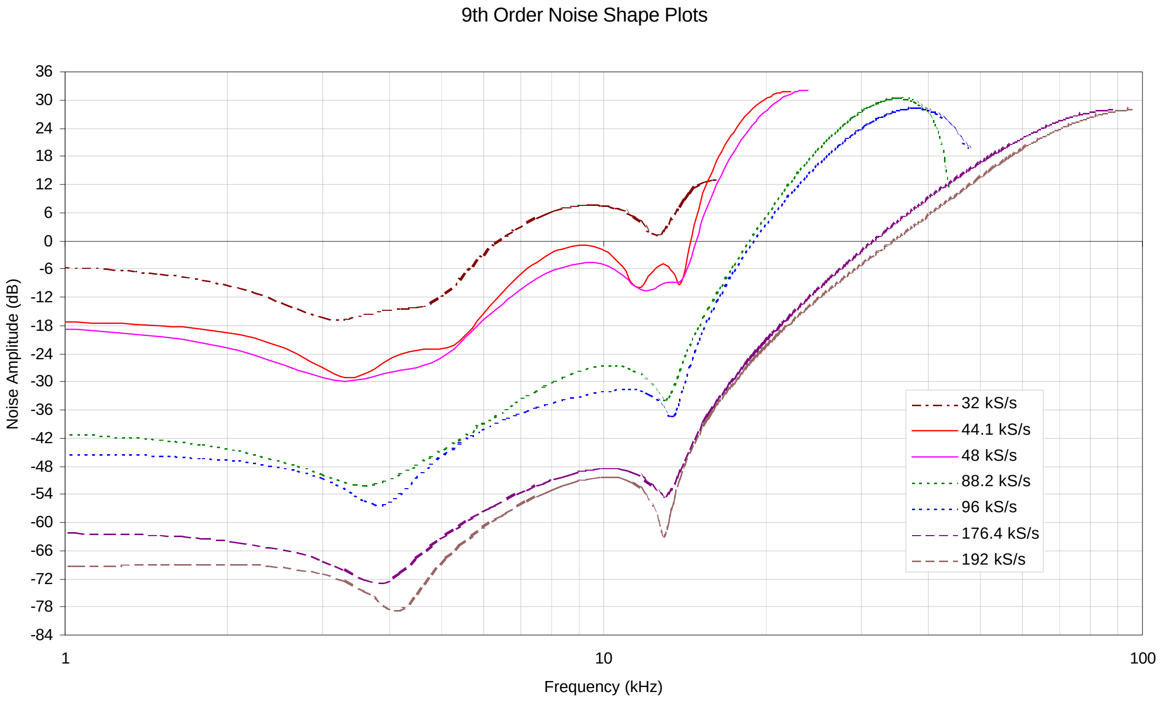

Figure 21 – 9th Order Noise Shapers implemented on ***dCS 90X***

The noise shaper plots above are all on the same vertical scales for easy comparison, and the vertical grid is approximately 1 bit per grid line. Note that for the audio band, 9th order noise shaping at 176.4 kS/s or 192 kS/s gives huge gains (8 bits or more).

This means that recording these formats on 8 channel 44.1 kS/s or 48 kS/s recording machines that store only 16 bits is quite practical, and there is, in practice, very little quality loss.

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	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 1	BNC x 3	
Connections	CH1 (left)		
	CH2 (right)		
	Word Clock In & Out		

Table 5 – SDIF-2 i/o specifications

SPDIF (electrical)	Output	
Type	*Single ended, ground referred*	
Impedance	75	Ω
Sensitivity (unloaded)	1.0	V pk-pk
Maximum Wordlength	24	bits
Damage level	> 10	V pk-pk
Connector	RCA Phono	

Table 6 – SPDIF (electrical) output specifications

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

Table 7 – SPDIF (optical) output specifications

Remote control interface	Input / Output	
Type	RS-232	
Level	RS-232	
Data Format	Contact ***dCS***	
Connector	9 way D-type male	

Table 8 - Remote Control Interface Details

Analogue Input Specifications

Balanced Inputs			
Type		Balanced	
Format		AES14 : 1992	
Impedance	+	5	kΩ
	-	5	kΩ
CMRR	50 Hz	>100	dB, spec
	50 Hz	>120	dB, typ
	1 kHz	>108	dB, typ
	10 kHz	>80	dB, typ
	20 kHz	>74	dB, typ
Level for Full Scale (as shipped)		+20	dBu
Trim range		±6	dB
Connector type		XLR3 female	
Connections	Pin 1	Ground or shield	
	Pin 2	+Signal	
	Pin 3	-Signal	

Table 9 - Analogue XLR Interface Details

The analogue inputs are balanced (not floating) with a stable, high common mode rejection ratio. Either input may be used on its own with the other floating if single ended operation is wanted.

AES3 (AES/EBU) Format

Message Handling

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

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

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

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

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

For more information on the way ***dCS*** implement the AES3 system message to handle higher sample rates, see the Appendix to this manual. For the formal definition of the AES3 interface, see footnote[8], from the AES.

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

How Far will it Go?

The AES/EBU format was designed to go reasonable distances, at 44.1 kS/s and 48 kS/s. Figure 22 and Figure 23 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 nsecs/metre.

At 96 kS/s (twice the data rate the format was designed for) the allowed cable length is less. Figure 24 and Figure 25 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.

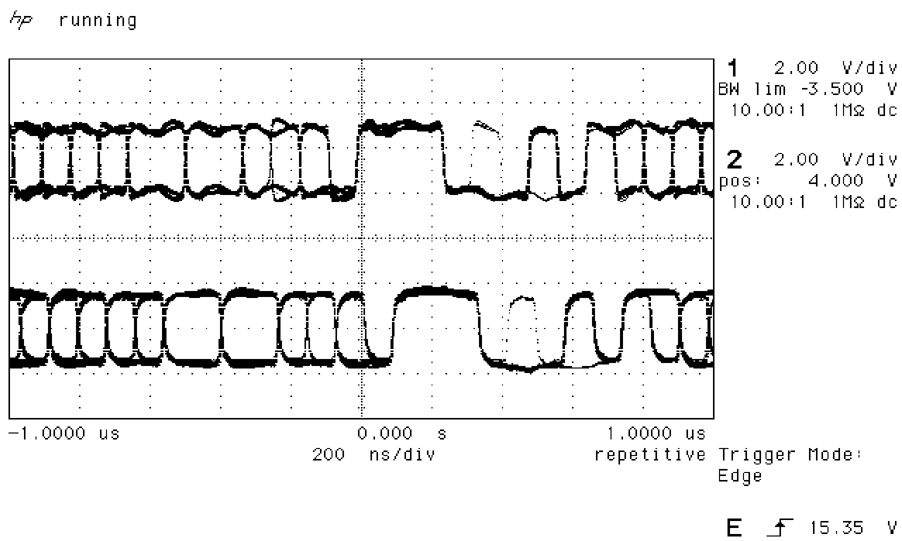

Figure 22 – AES3 format at 48 kS/s over 16 metres

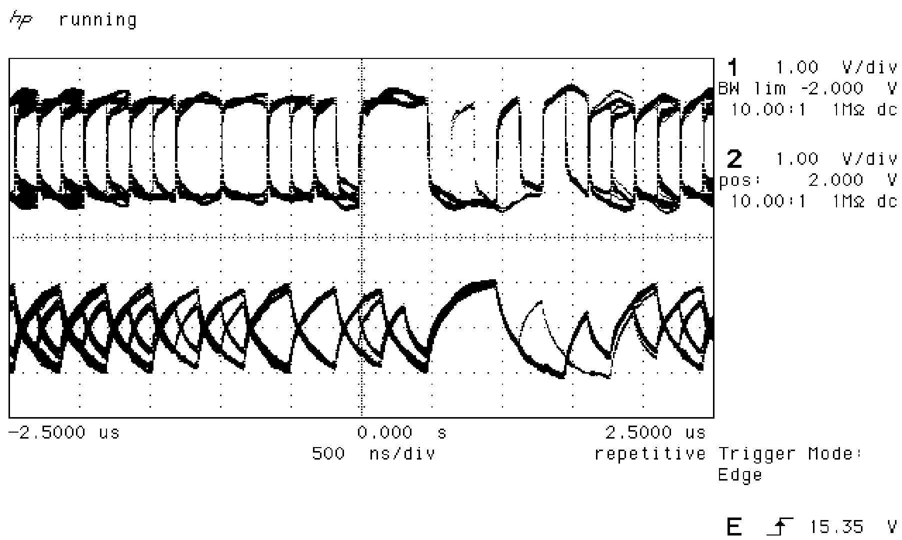

Figure 23 – AES3 format at 48 kS/s over 94 metres

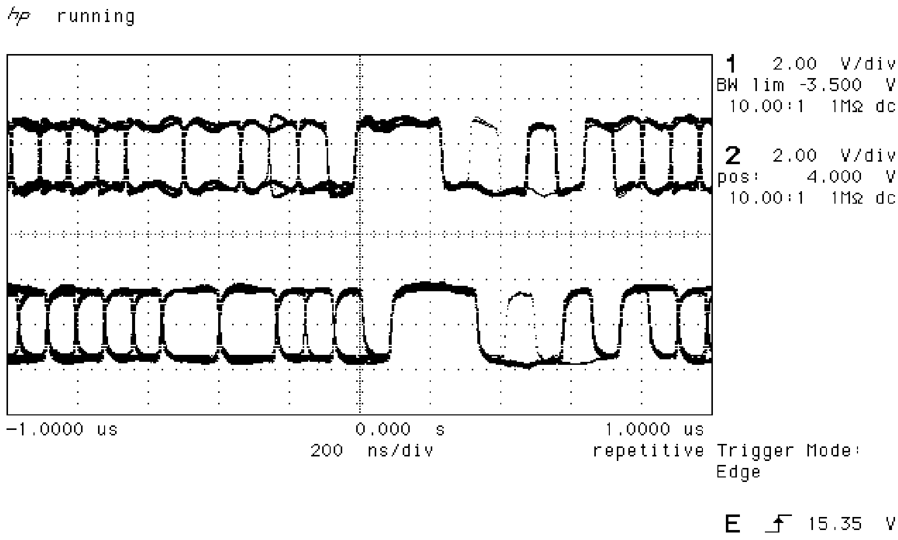

Figure 24 – AES3 format at 96 kS/s over 16 metres

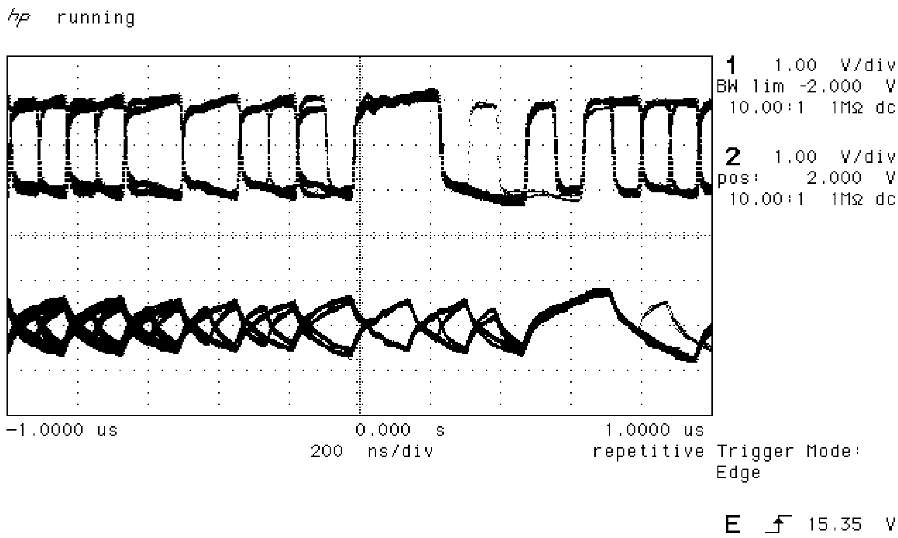

Figure 25 – AES3 format at 96 kS/s over 94 metres

SPDIF

Message Handling

The SPDIF interface (sometimes known as the Consumer AES/EBU interface) transmits a data structure that conforms to the IEC 958[9] standard. Like the AES/EBU, this contains 28 bits of Manchester encoded data, and a 4 bit near-Manchester encoded preamble in a subframe, and subframes are further assembled in a block and frame structure. The difference lies only in the voltage levels, and the Channel Status bits (the System Message). It contains 24 bits of audio data.

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

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

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

[9] See EN 60958:1995 or IEC958:1989 with amendments 1 & 2. The structure of the message is sufficiently complex that it is best to read the source material.

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
Right Channel
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 are SDIF-2 waveforms (data and word clock) at 96kS/s and 44.1kS/s.

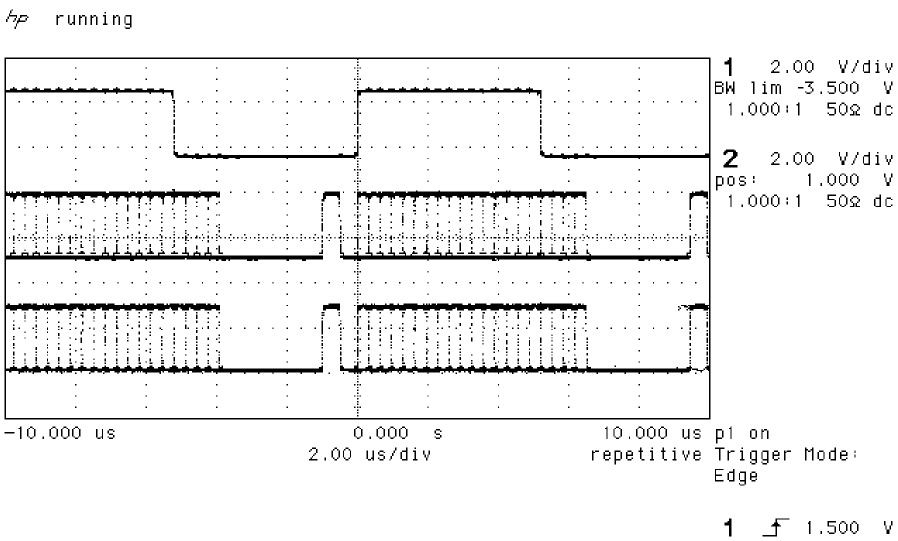

Figure 26 – SDIF-2 PCM format at 96 kS/s

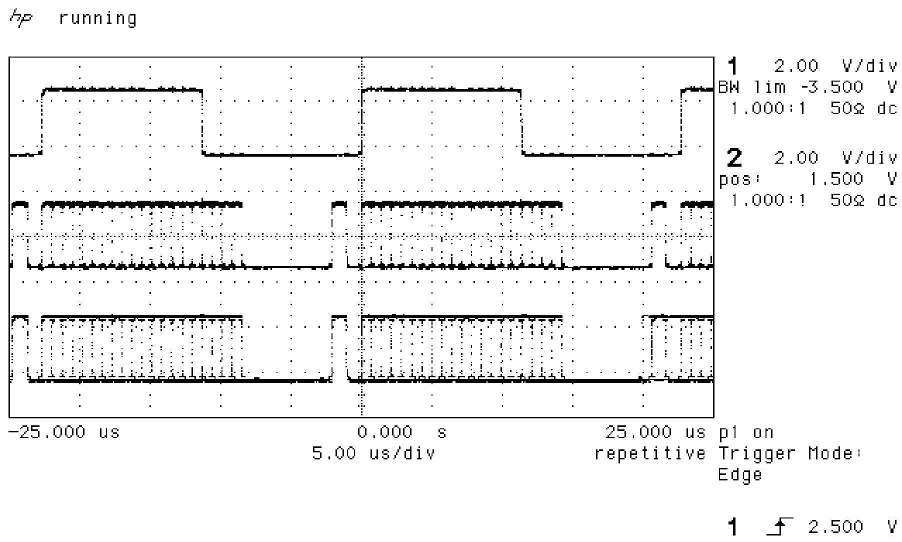

Figure 27 – SDIF-2 PCM format at 44.1 kS/s

SDIF-2 Messaging

The SDIF-2 message is given in the table following. The ***dCS 900*** & ***902*** implementation sets all bits of the User message to "0".

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 10 – SDIF-2 Message Table

Power Consumption

The ***dCS 900*** & ***902*** have 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. Consumption is independent of mains voltage selector switch setting.

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

Nominal mains	25 W
Mains -10%	22 W
Mains +10%	27 W

The actual intended supply voltage is shown on the rear panel. 50Hz or 60Hz operation is not important – the unit can use either. In general, users will not need to change the mains input configuration. If you do need this to be done, please see the section **"Having Your Options Changed"**, page 54 in this manual and contact your distributor or ***dCS***.

Size, Weight and Operating Conditions

Size and Weight

The ***dCS 900*** & ***902*** 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 case of the ***dCS 900*** & ***902*** have no ventilation slots or fan cooling, to give:

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

It dissipates relatively low power, so that usually allowing natural convection provides enough cooling. Do not install the unit near heat sources such as radiators, air ducts or direct strong sunlight. Ambient should not exceed 50°C, should not fall below 0°C, and should be a non condensing. If in doubt, the easy test is – the ***dCS 900*** & ***902*** are happy to work anywhere a human is.

GENERAL TECHNICAL INFORMATION

Word Length Reduction

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

There is really only one way of tackling the problem – another signal has to be added to the wanted one to smooth the staircase transfer function that truncation causes. Mathematically, with two signals present, the transfer function that the wanted signal sees is the convolution of the PDF[10] of the second signal and the staircase function. The converse is also true – the transfer function the additional signal sees is the convolution of the PDF of the wanted signal and the staircase function. This aspect is not a problem with the dither types considered below, but it can be with some highly frequency shaped dithers.

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

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

The table below gives the actual noise levels for 16 bit truncated signals with no dither, various types of dither, noise shaping on its own, and noise shaping with dither. The 0 dB reference level is taken as the minimum noise we could possibly get away with – the amount that simple 16 bit truncation (16 bit Q noise) would give, if it were well behaved, which it is not.

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

Straight forward dither always adds noise – it can only produce signals with a noise floor higher than Q noise on its own. However, the noise power added is a few dBs for simple types. Noise shaping adds rather more noise, but it can be made to add it in parts of the spectrum that the ear is less sensitive to, so the perceived noise (F weighted noise) is lower – up to three bits lower. It results in a signal that the ear hears as having a far **lower** noise floor than a 16 bit truncated signal, rather than the "not much worse" of dither alone, even though there is really more noise present[11].

Truncation Type, with 44.1 kS/s data rate	**Noise, unweighted, rel 16 bit Q noise[12]**	**Noise, F weighted, rel 16 bit Q noise**	**Comments**
16 bit truncation	0 dB	0 dB	Unpleasant low level effects
16 bit truncation with Top Hat dither	3 dB	3 dB	Okay – can show noise modulation at low signal levels
16 bit truncation with Triangular dither	4.8 dB	4.8 dB	All noise modulation and unpleasant effects removed, but noise floor is high
16 bit truncation with Noise Shaped Triangular dither	4.8 dB	1.2 dB	All noise modulation and unpleasant effects removed. Not much perceived noise penalty
16 bit truncation with 3rd order noise shaping and no dither	6.9 dB	-10.5 dB	Okay with input noise floors down to –102 dB
16 bit truncation with 3rd order noise shaping and Noise Shaped Triangular dither	11.0 dB	-9.2 dB	Unconditionally free from truncation effects with all inputs
16 bit truncation with 9th order noise shaping and no dither	23.4 dB	-17.9 dB	Okay with input noise floors down to –120 dB
16 bit truncation with 9th order noise shaping and Noise Shaped Triangular dither	28.2 dB	-16.7 dB	Unconditionally free from truncation effects with all inputs

Table 11 – Dither and Noise Shaping Noise Powers

Noise shaping on its own is not perfect. It relies on a small amount of noise in the input signal to generate the frequency shaped correction signal, and if there is very low noise in the input signal, this mechanism can break down. With ADCs, however, this situation does not arise, because of the analogue noise in the ADC and the input signal.

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

[12] 16 bit Q noise is -98.1 dB relative to a full scale sine wave.

There is another option not supported by the ***dCS 900** & **902*** – generate the dither independently of the signal and frequency shape it prior to addition, but do not add it in an error shaping loop. This seems to ***dCS*** to combine the worst of all worlds – the high noise floor in the 0-6 kHz area of straight dither, and the high total noise of noise shaping. However, some people use it.

What does it look like?

Figure 28 gives the spectra of 16 bit truncated 44.1 kS/s signals with a –90dB sine present, for two dither only signals (Top Hat, Noise Shaped Triangular), and with a 10th order noise shaped[13] signal, generated and processed by a ***dCS 972***. The equivalent simply truncated spectrum is shown in Figure 29, separately because it is so revolting. In it, we can see that at the signal level shown (-90 dB) error power from the quantising/truncation is beginning to pile into the fundamental, which is showing an amplitude error of +1.3 dB, as well as all the unwanted harmonics. This would show up on a conventional linearity plot, although the sign of the error could be either way.

We see that the noise shaping approach maintains low noise in the critical audio mid band.

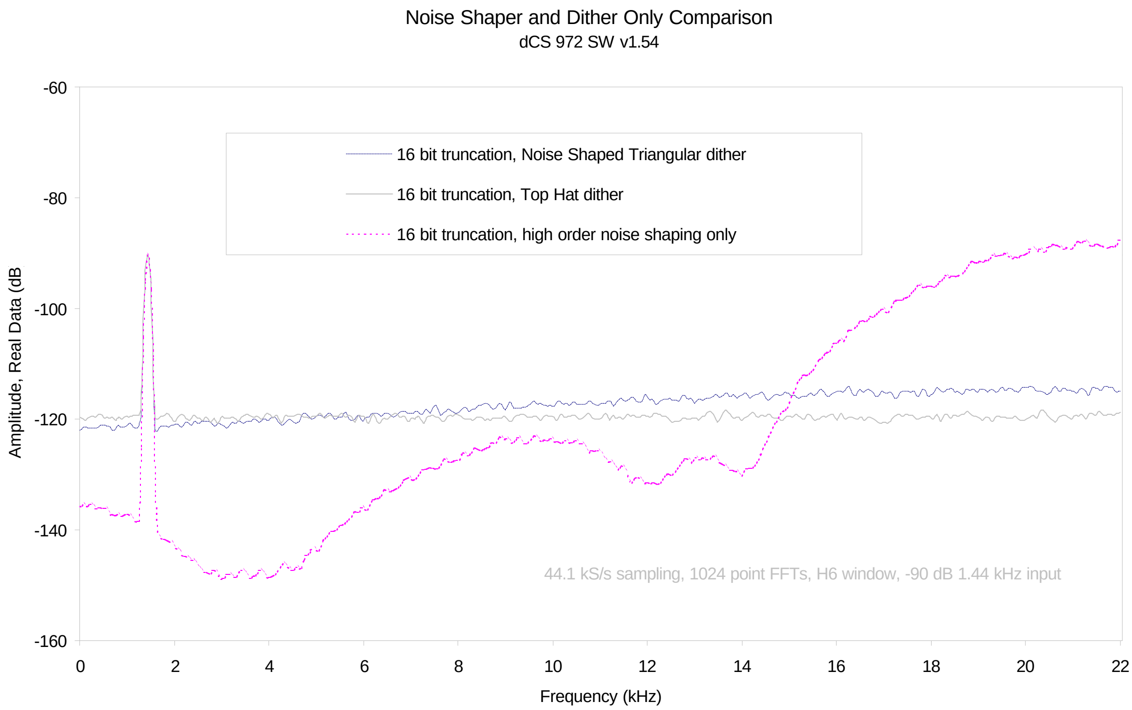

Figure 28 – Noise Shaping and Dither Spectra

13 for comparison with the table, 10th and 9th order noise shaping are very similar.

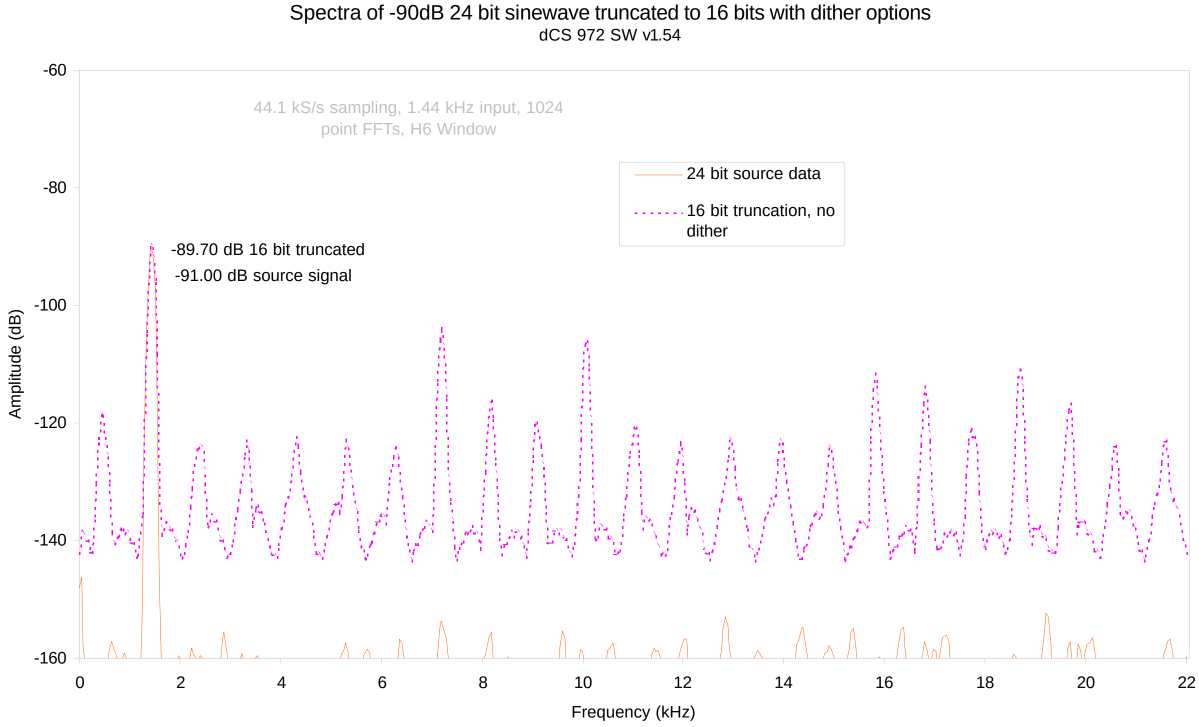

Figure 29 – Truncation Only Spectra

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 44.056 kS/s and 47.952 kS/s). These must be 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 900 for use in <*country*>, with options ….

or

dCS 902 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 900*** & ***902*** 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 (150 mA at 110 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.

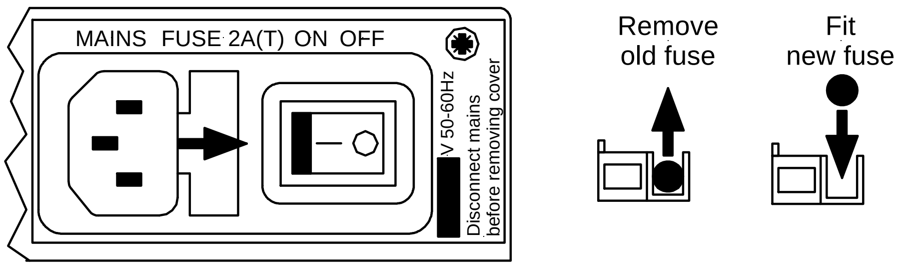

Figure 30 – Changing the Mains Fuse

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

Software

Installing New Software

The operating software can be downloaded via the RS-232 link from a PC com port, using the Windows Remote software running on the PC.

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

To find out if there are any software updates available for your equipment, call us, or email us, with your unit's serial number.

Hardware Update or Calibration

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

Safety and Electrical Safety

There are no user serviceable parts inside the ***dCS 900*** & ***902*** 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 900*** & ***902***.

TROUBLESHOOTING

Error Codes and Messages

The error codes reported by ***dCS 900*** & ***902*** 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 12 - 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 sample rate is changed.
Hot	The unit is overheating, and performance may suffer.
Ouch	The unit is seriously overheating, and may be damaged shortly. Switch off!
Bad Fs	The sample rate coming in is not one the unit can lock to, or there is an input signal quality problem.

Table 13 - System Error Codes

Trouble Shooting Your System

If you experience difficulties when using your ***dCS 900 / 902***, 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 56.
- Check that the mains cable is pushed fully home into the mains inlet in the rear of the unit.

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 equipment are switched on and correctly set up.
- Check that an audio signal is present on one or both of the inputs.
- Ensure **Mute** is not enabled - LED off.

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

- Ensure the trim tool or 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 unit slaves to Word Clock but not AES/EBU

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

The Left and Right channels are swapped

- Check that the audio input cables are not reversed.
- Check that the channels are not swapped elsewhere in the system.
- If using **SDIF-2**, check that **CH1** and **CH2** outputs are connected correctly.
- *dCS 902* only: In **Dual AES** mode, ensure that the **AES 1** output is connected to the input on the destination equipment for the Left channel data (probably labelled AES 1, AES A or Left) and **AES 2** output is connected to the input on the destination equipment for the Right channel data (probably labelled AES 2, AES B or Right). See the manual of the destination equipment for information.

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.

Clicks or crackles occur on the outputs

- Check that all cables are connected correctly and not damaged.
- Check that the overload LED does not light on signal peaks.

Periodic clicks occur in the recording

- If you are using a Master Clock, check that the ADC AND the recorder are sync'ed to it.

The unit fails to slave to a Master Clock

- Press the **Master/Slave** button to select Slave mode. If a suitable reference is connected, the LED should light and the unit should lock after a few seconds.
- Check that the **Reference In** or **Clk 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 Overload indicator will not go out

- Remove any input and reference input. If the condition persists, contact your dealer or *dCS* - the unit may be faulty.

External meter does not show overload, *dCS 900 / 902* does.

- The *dCS 900 / 902* monitors a number of internal points and data word values in its calculation of overload. This may cause it to indicate an overload while the final external data word may not show it - for example with a very narrow but large spike, that the low pass decimation filter may broaden out sufficiently that the output data does not saturate. It is likely that if the *dCS 900 / 902* says it is in overload, it is. You can choose to ignore it!
- Some digital meters are quite insensitive to overloads. Such equipment may include a sensitivity setting, where an overload is only flagged when a number of consecutive digital words saturate - typically 1, 2, 4 or 8 consecutive samples. There is some justification for this - single saturation events are not always audible. The *dCS 900 / 902* flags their presence - it is up to the recording engineer to decide what to do about it.

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 14 – *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 software versions 1.3x to1.5x.
v1.3x changes: Word Clock & AES Reference synchronisation improved.
v1.5x changes: 904 related enhancements and hardware harmonisation.
v1.5x versions differ only in a few minor bug fixes.

Definitions of Units

dB A relative level in decibels. The context should state the reference level.

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. Sometimes referred to as an A/D Converter.

DAC Digital to Analogue Converter. Sometimes referred to as a D/A Converter.

DDC Digital to Digital Converter. Sometimes referred to as a D/D Converter.

Full Contents

Tables

Figures

Keywords and Phrases