dCS 955 Stereo Digital to Analogue Converter

User Manual

Software Release 1.0x

June 2002

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

PRODUCT OVERVIEW

The *dCS 955* 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, 192kS/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 multichannel 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. Common setups may be stored and re-loaded later. 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

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.

Product Features

Formats

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

Syncing

- Can sync to Word Clock, AES Reference or input signal.
- 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 (88.2kS/s, 96kS/s)
- Dual or Quad AES (176.4kS/s and 192kS/s)
- DDC mode converts Dual AES or Quad AES to High speed Single AES.
- DDC mode converts 4-wire DSD to SDIF DSD format.
- Digital Volume control (PCM inputs only).

Metering

• Flexible metering with average or peak characteristics, peak hold function.

Test Generator

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

Ease of Use

- Remembers last settings.
- 6 user-defined set ups and up to 12 preset set ups.
- 3 user-defined Function buttons.
- Auto-detects sample rate, PCM input format and SDIF input format.
- Front panel lockout prevents unauthorised changes.
- Software upgrade-able without opening the box, from PC or CD-ROM.
- Can be remote controlled from a PC.

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About this Manual

If you have not used a dCS 955 before, please read the section "Using your dCS 955 for the first time" on page 88.

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

- table of contents (page 5)
- step-by-step (page **10**) and applications guides (page **18**)
- detailed software and hardware information (page **26**)
- technical information (page **58**)
- information for first time users (page **88**)
- options, maintenance and troubleshooting (page **90**)
- index section (page 104)

What does the coloured text mean?

If you are reading a colour print or a soft copy of this manual, you will notice that some types of text are in colour:

- **Brown text in bold** is a reference to another section or page. Sometimes, if you are reading a soft copy of the manual, page numbers are hyperlinks click on them and you will go there.
- Blue text, White text on black or Red text outlined is used for controls and connectors, to distinguish between alternative functions.
- Pink text is a menu page or setting.
- Green text in bold shows what appears on the display.
- Purple text in bold is used for indicators.

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

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

About Sample Rates

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

STEP-BY-STEP GUIDE

This section guides you through setting up the unit for basic operation. You may find this useful if you have not used the dCS 955 for a while.

Preliminaries

The **Menu Guide** sheet details the menu structure. For more information, see the Menu section, starting on page **26**.

Cables

Connect up with cables designed for digital audio:

- for AES/EBU interfaces use 110Ω screened, twisted pair cables fitted with one male XLR connector and one female XLR connector.
- for DSD/SDIF or Word Clock interfaces, use 75 Ω coax cables fitted with BNC plugs.

For analogue outputs, use with screened cables of the correct type:

- for balanced outputs, use screened, twisted pair cables fitted with one male XLR connector and one female XLR connector.
- for unbalanced outputs, use coax cables fitted with RCA Phono plugs.

Powering Up

Connect power to the unit. If there is a power switch next to the power inlet on the rear panel, ensure it is set to on. Press the **On/Off** button and wait for about 20 seconds while the unit configures itself. The display will show dCS955 **Digital to Analogue Converter**, followed by **Please wait** ... When the unit has booted up, either the meter display or status display will appear:

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Loading a Default Set-up

The following routine sets the unit to a defined state – you will not have to do this every time you power up as the unit remembers the last set-up before power-down.

do this: To open the menu, press the Menu button. Press the ↑ button then the OK button twice to open the Recall Settings menu. Press ↓ then OK to open the Fixed Stores menu, then press the OK button to load Store A. The display will prompt you to press one of the F buttons to confirm the action. Wait a few seconds for the settings to be loaded, then hold the Cancel button for few seconds to exit the menu.

Step 1 – Setting the Input Format



Figure 1 – Input Format Selection

do this: Press Menu then OK to select the DAC Settings menu. Press OK to select the Input Format menu. Press ↑ or ↓ to move the highlight to the required format then press OK to select it. Press Cancel repeatedly to exit the menu or wait for timeout.

The most common format is PCM. Do not set to DSD or DSD P3D unless your source generates DSD.

Step 2 - Selecting an Input

Choose one of the following five section	IS:
--	-----

Connecting to a PCM Single AES source

do this: do this:	Connect the single AES output on your source equipment to the matching input on the <i>955</i> rear panel using suitable cables. An AES3 source (XLR connector) may be connected to any of the four AES/EBU inputs. Press the corresponding AES button.
	If the source is generating digital data, the unit will lock, display the sample rate / word length and un-mute. The indicator over the input button will turn green.
do this:	Proceed to Step 3.

Connecting to a PCM Dual AES Source

- do this: Check that your source equipment is capable of Dual AES operation.
- do this: Connect the AES 1 (or AES A) output on your source equipment to the AES 1 input on the 955 rear panel and the AES 2 (or AES B) output to the AES 2 input, using two XLR cables. Ensure the cables are not swapped.
- do this: Press the AES 1 button repeatedly until the display changes to AES1-2 (Dual Wire).
- do this: Alternatively, connect to the AES 3 and AES 4 inputs and press the AES 3 button repeatedly until the display changes to AES3-4 (Dual Wire).

If the source is generating digital data, the unit will lock, display the sample rate / word length and un-mute. The indicators over the two selected **AES** buttons will turn green.

do this: Proceed to Step 3.

Connecting to a DSD P3D source

- do this: Check that your source equipment is capable of DSD P3D operation.
- do this: Connect the AES 1 output on your source equipment to the AES 1 input on the 955 rear panel, the AES 2 output to the AES 2 input and the AES 3 output to the AES 3 input, using three XLR cables. Ensure the cables are connected in the correct order.
- do this: Press any AES button the display will show AES packed DSD.

If the source is generating digital data, the unit will lock, display **DSD** / selected input and un-mute. The indicators over the three **AES** buttons will turn green.

do this: Proceed to Step 3.

Connecting to a PCM Quad AES or DSD Quad source

- do this: Check that your source equipment is capable of Quad AES or DSD Quad operation.
- do this: Connect the AES 1 output on your source equipment to the AES 1 input on the 955 rear panel, the AES 2 output to the AES 2 input, the AES 3 output to the AES 3 input and the AES 4 output to the AES 4 input, using four XLR cables. Ensure the cables are connected in the correct order.
- do this: For PCM format, press the AES 1 button repeatedly until the display changes to AES1-4 (Quad Wire).
- do this: For DSD format, press any **AES** button the display will show **AES** packed **DSD**.

If the source is generating digital data, the unit will lock, display the sample rate (or **DSD**) / selected input and un-mute. The indicators over the four **AES** buttons will turn green.

do this: Proceed to Step 3.

Connecting to a PCM SDIF-2 or DSD SDIF-2 source

- do this: If the Input Format is set to DSD, check that your source equipment is capable of DSD-SDIF operation.
- do this: Connect the SDIF output on your source equipment to the DSD/SDIF connectors on the 955 rear panel using three coax cables. Connect CH1 out to CH1, CH2 out to CH2 and CLK out to CLK IN.

do this: Press the **BNC** button - the display will show **BNC packed DSD**.

If the source is generating digital data, the unit will lock, display **DSD** / selected input and un-mute. The indicator over the **BNC** button will turn green.

The 955 will automatically detect either SDIF-2 or SDIF-3.

do this: Proceed to Step 3.

Step 3 – Connecting the Analogue Outputs

The **Balanced Outputs** and **Unbalanced Outputs** are separately buffered. Both pairs of outputs may be connected at the same time, if required.

Choose one of the following two sections:

Using the Balanced Outputs

do this: Use the **Balanced Outputs** if you want to connect *955* to a device with balanced inputs only. Connect to the destination with two screened XLR cables.

IMPORTANT! The balanced outputs are semi-floating. Do not try to connect one side to an unbalanced input as this will prevent the output stage working properly.

do this: If you wish, you can adjust the balanced output full scale levels using the two **Sensitivity** trimmers mounted on the rear panel. Use a small flat-bladed screwdriver or trim tool. Take care to keep the two channels balanced in level.

If the source is playing, you should have audio. In PCM mode, you can adjust the Volume using the \uparrow and \downarrow buttons.

do this: Proceed to Step 4.

Using the Unbalanced Outputs

do this: Connect the Unbalanced Outputs to a device with unbalanced inputs only. Use two screened coax cables fitted with phono connectors.

The full scale levels at the **Unbalanced Outputs** are fixed.

If the source is playing, you should have audio. In PCM mode, you can adjust the Volume using the \uparrow and \downarrow buttons.

do this: Proceed to Step 4.

Step 4 – Setting the Sync Source

do this: The 955 will be set to sync to the selected Audio Input. If you do not want to use an external reference clock, proceed to Other Settings.



Figure 2 – Sync Source Selection

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

Syncing to an External Wordclock

If you want to synchronise your system to Wordclock from a Master Clock (such as a *dCS 995* or *dCS 992*) or other stable source, do the following:

- do this: Set the Master Clock sample rate to match the source (probably 44.1 or 48kS/s).
- do this: Connect either a Wordclock or AES/EBU output from the Master Clock to the clock input on the source equipment and ensure it is locked.
- do this: Connect another Wordclock output from the Master Clock to the CLK IN connector on the 955 rear panel.
- do this: If the source equipment uses SDIF (in either PCM or DSD mode), the Wordclock feed from the Master Clock replaces the Wordclock feed from the source equipment.
- do this: Press Menu and use the ↓ button to scroll down the top level to the Connector Settings menu. Press OK twice to open the Sync Mode menu. Use the ↓ button to scroll down to Slave to word Clock and press OK.

The unit will re-lock.

do this: Proceed to Other Settings.

Syncing to an AES/EBU Reference

If you want to synchronise your system to an AES/EBU Reference from a Master Clock (such as a *dCS* 995 or *dCS* 992) or other stable source, do the following:

- do this: Set the Master Clock sample rate to match the source (probably 44.1 or 48kS/s).
- do this: Connect either an AES/EBU or Word Clock output from the Master Clock to the clock input on the source equipment and ensure it is locked.
- do this: Connect another AES/EBU output from the Master Clock to the **Reference In** connector on the *955* rear panel.
- do this: Press Menu and use the \downarrow button to scroll down the top level to the Connector Settings menu. Press OK twice to open the Sync Mode menu. Use the \downarrow button to scroll down to Slave to AES Reference and press OK.
- do this: Press **Cancel** twice to back up to **Connector** Settings menu, press the \downarrow button, then **OK** twice to open the Reference Mode menu. If you are not looping the AES Reference through the unit, use the \downarrow button to scroll down to Loop Terminated, otherwise scroll down to Loop Through. Press **OK**.

The unit will re-lock.

do this: Proceed to Other Settings.

Other Settings

The basic set-up procedure is complete.

Many more features are available through the menu. See the menu section, starting on page **26** for more information.

TYPICAL APPLICATIONS

Using a *dCS* 955 with SDIF DSD



Figure 3 – SDIF-DSD input configuration

do this:	Set the Input Format menu to DSD.
do this:	Select a filter.
do this:	Ensure Mute is off.

Using a Master Clock to Sync a *dCS* 955



Figure 4 – Syncing a 955 to a Master Clock

clock.
urce to it.
-

Replaying Recorded DSD from an 8-track 16/44.1 PCM Recorder



Figure 5 – Replaying recorded 2 channel DSD from an 8-track 16/44.1 PCM recorder

do this:	Set the Input Format menu to DSD.
do this:	Connect the 4-wire DSD source, ensuring the wires are in the right order.
do this:	Press the AES 1 button repeatedly until AES1-4 (Quad Wire) is displayed.
do this:	Select a filter and ensure Mute is off.

Operating Several Units on One Remote Chain



Figure 6 – Multi-unit Remote Control daisy chain

A PC running the Windows Remote Control program can control a mixture of dCS units (up to about 10) on each daisy chain. To make them individually addressable, each unit must be set to a different RS232 Address. They can then be identified, and grouped, in the remote window.

See "Remote In & Out" on page **56** for cable details and "RS-232 Remote Control Interface" on page **77** for more details.

Six Channel PCM Set Up





- do this: The top unit must have its AES Settings / Ref Mode menu set to Slave to Audio Input, the middle unit to Loop Through and the bottom unit to Loop Terminated. For all units, set the Sync Source / Auto Slave menu to On.
- do this: Alternatively, Word Clock may be used, setting the Sync Source / Sync Mode menu on all units to Slave to Word Clock.

The units self align quite accurately (see the "Sample Alignment" section on page **65** onwards).

If the six channels are not bit-aligned, all three units should have the Sync Source / Sync Mode menu on all units to Audio Input.



Replaying Recorded 6 Channel DSD from a 24-Track 16/44.1 PCM Recorder

Figure 8 – Replaying a 6 channel DSD recording from a 24-track 16/44.1 recorder

do this:	The top unit must have its AES Settings / Ref Mode menu set to Slave to
	Audio Input, the middle unit to Loop Through and the bottom unit to Loop
	Terminated. For all units, set the Sync Source / Auto Slave menu to On.
do this:	Alternatively, Word Clock may be used, setting the Sync Source / Sync
	Mode menu on all units to Slave to Word Clock.
do this:	For all three units, set the Input Format menu to DSD and press one of the AES
	buttons to select the 4-wire DSD input.
do this:	Take care to ensure the input wires are connected in the right order.

The units self align quite accurately (see the "Sample Alignment" section on page **65** onwards).

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

Replaying 8 Channel P3D DSD



Figure 9 – 8 channel P3D DSD set up

Four *955* units and a Master Clock can be used as above to convert 8 channels of P3D encoded DSD. Note that a 24-bit recorder is needed for P3D.

- do this: Set the Master Clock to 44.1kHz. do this: For all four DACs, set the Input Format menu to DS
 - is: For all four DACs, set the Input Format menu to DSD (P3D Mode), press an AES button to select the P3D input. Select a filter and ensure Mute is off.

Upsampling a CD



Figure 10 – Upsampling a CD to 24 bit / 192kS/s

do this: Set the *dCS* 974 Input Sample Rate to Auto, Output Sample Rate to 192ks/s, Output Mode to Dual AES, Sync Source to Word Clock.
do this: On the 955, press the AES 1 button repeatedly to set AES1-2 (Dual Wire), ensure Mute is off and select a filter. Set Sync Mode to Slave to Word

The Master Clock is optional – it helps reduce jitter.

- do this: Set the Master Clock to generate 44.1 AND 96kS/s simultaneously.
- do this: If you do not have a Master Clock, you can slave the *dCS* 974 to the CD player by setting Sync Source to Audio Input and slave the 955 to the *dCS* 974 by setting Sync Mode to Audio Input.

clock.

Converting Quad AES to Single AES



Figure 11 – Converting Quad AES to double speed Single AES

do this: Press the AES 1 button repeatedly until AES1-4 (Quad Wire) is displayed, set the DDC Mode menu to On.

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 12 – Converting 4-wire DSD to SDIF DSD

do this:	Set the Input Format to DSD and press the AES 1 button to select DSD Quad wire operation.
do this:	Set the DDC Mode menu to On.
do this:	Select a filter and ensure Mute is off.
	This arrangement can also be used with a P3D source.
do this:	Connect the P3D source to inputs AES 1 – 3 .
do this:	Set the Input Format to P3D and press the AES 1 button.

Replaying 24/192 from 2 Nagra-D Recorders



Figure 13 – 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 *955* allows for small timing discrepancies between the **AES 1-2** pair and the **AES 3-4** pair.

THE SOFTWARE - THE MENU



Figure 14 – Menu flow chart

Using the Menu

The menu gives the user access to a wide range of additional features. It also allows new features and performance enhancements to be added at a later date by software upgrades.

Opening the Menu

The Menu is controlled by the four **Menu Operation** buttons:

- the Menu / OK button opens the menu and selects pages or options.
- the \uparrow and \downarrow buttons page backwards and forwards through the menu.
- the **Cancel** button navigates back towards the top level menu.

You cannot go directly to any particular page, but must enter at the top of the menu and then navigate through the menus until you reach the page you want.

Navigating Through the Menu

The menu is arranged as a series of levels, each with a index number.

do this: Press the Menu button, the screen changes to:



do this: Press the \downarrow button to scroll down to the second page of the top level menu:

do this:

Press the **OK** button to select this and the menu will indent to the second level:



The selected menu page is highlighted (2.1 Swap L/R Channels). The numbering (2.1) indicates a second level menu page.

do this: Use the \uparrow and \downarrow buttons to scroll round the list.



do this:

Scroll back to highlight menu page 2.1 and press the **OK** button, the display changes to:



The arrow symbol in the top left corner indicates that you have reached the bottom level and the list is of items you can select, rather than another page. The * indicates the currently selected item.

do this: Use the ↓ button to move the highlight to On and press the OK button. The display changes to:



The channels are now swapped.

Menu options that are not available (usually due to other settings, particularly DSD modes) disappear from the list. When the happens, the menu numbering may jump from (for example) **1.3** Filter Settings to **1.5** DDC Mode.

Closing the Menu

There are two ways to close the menu and return to normal operation. The easiest way is to wait 15 seconds for the unit to time-out and revert to the standard display. Alternatively, hold down the **Cancel** button for a few seconds to return to the status or meter display.

If the unit times out before the operation in hand has been completed, simply reenter the menu, page forward (or backward) and continue where you left off. If you find the 15 second time-out difficult to use, you can extend it or disable it by changing the Display Timeout setting in the Display menu.

do this: For practice, open the menu again and set the Swap page to Off.

Menu Sequence

Use the flow chart (**Figure 14**) or the **Menu Guide** sheet to guide you through the menu more quickly. When you are more familiar with the menu, you will find it more convenient to perform all the operations in one go before finally closing the menu.

The following sections detail the top level menu, then each second level menu in turn.

Top Level Menu

The top level menu groups the menu features conveniently, reducing the menu navigation time.



Sets some of the primary operating parameters of the DAC.



Sets some of the secondary operating parameters of the DAC.



Sets display-related parameters.



Sets the connector interface formats.



Sets the meter type and characteristics.



Sets up the test tone generator and runs other test routines.



Displays unit and contact information.



Save and recall unit settings.

1. DAC Settings Menu



Figure 15 – DAC settings menu

1.1 Input Format

Sets the format of the input data. The options are:

РСМ	Decodes linear PCM data as either single-wire,
	Dual-wire or Quad-wire AES on the AES 1 - 4
	inputs or SDIF-2 on the DSD/SDIF interface.
DSD	Decodes DSD data on the AES 1 - 4 Quad-wire
	input (AES packed DSD) or the DSD/SDIF
	interface (BNC packed DSD).
P3D	Decodes DSD data on the AES 1 - 3 Triple-wire
	input (AES packed DSD) or the DSD/SDIF
	interface (BNC packed DSD).

1.2 Input Selection

Selects the active input. The options are:

AES	1,2,3 or 4	Single-wire AES3 at up to 96kS/s.
AES	1 - 2	Dual-wire AES3 on the AES 1 + AES 2 input at
		88.2, 96, 176.4 or 192kS/s.
AES	3 – 4	Dual-wire AES3 on the AES 3 + AES 4 input at
		88.2, 96, 176.4 or 192kS/s.
AES	1 - 4	Quad-wire AES3 or DSD on the AES 1 - 4 input at
		176.4 or 192kS/s.
BNC		SDIF-2 on the DSD/SDIF interface at up to 96kS/s,
		or DSD/SDIF-2 or DSD/SDIF-3.

In DSD mode, the choices are reduced to AES $\,1$ (DSD Quad or P3D) or BNC (SDIF DSD).

It is usually more convenient to select the active input with the front panel controls.

1.3 Filter Settings

Selects an optional anti-image filter.

For PCM mode, these offer different trade-offs between Nyquist imaging and energy smear.

Filter 1	Symmetrical filter, with the time response before a transition the mirror image of the time response after. Offers the sharpest cut-off, minimum Nyquist imaging but longest energy smear.
Filter 2	Symmetrical filter with more relaxed cut-off, more
	Nyquist imaging and lower energy smear.
Filter 3	Symmetrical filter, more relaxed than Filter 2.
Filter 4	Symmetrical filter with very relaxed cut-off, poor
	Nyquist imaging but shortest energy smear.
Filter 5	176.4 or 192kS/s only: Gaussian filter.
Filter 6	176.4 or 192kS/s only: An asymmetrical filter,
	offering almost no time response before the initial
	step.

For DSD mode, there are four filters but these progressively reduce the out-ofband noise level. If your system gives undesirable effects with Filter 1, try Filter 2 then Filter 3.

Filter 1	Widest bandwidth but highest out-of band noise level.
Filter 2	Slightly lower bandwidth, less noise.
Filter 3	Lower bandwidth and less noise than Filter 2.
Filter 4	Minimum noise with 20kHz bandwidth, intended for metering applications.

1.4 De-Emphasis

In PCM mode at 32, 44.1 or 48kS/s only, sets the De-Emphasis characteristic applied to the input signal. If the De-Emphasis message in the data stream is wrong (the tonal balance will be wrong), correct this with one of the first 3 settings.

off	Applies no De-Emphasis and ignores the
	Emphasis message flag.
50/15us	Applies 50/15µs De-Emphasis curve and ignores
	the Emphasis message flag.
CCITT J17	Applies CCITT J17 De-Emphasis curve and
	ignores the Emphasis message flag.
Auto	Decodes the Emphasis message flag and applies the appropriate De-Emphasis curve. This is the usual setting.

1.5 DDC Mode

When set to On, performs sample rate and/or format conversion on the incoming data stream and outputs digital data. The conversion applied depends on the input format:

Single-wire AES	Copied to AES Reference Out with no conversion.
Dual- or Quad-	Converted to double speed single-wire AES and
wire AES	output on Reference Out. 88.2 or 176.4kS/s are
	converted to 88.2kS/s. 96 or 192kS/s are
	converted to 96kS/s.
PCM SDIF-2	Outputs AES clock with no data at the Wordclock
	rate on Reference Out.
4-wire DSD	Outputs SDIF format DSD on the SDIF inputs.
	SDIF CH1 is checked for activity first, before the
	SDIF inputs are changed to outputs.

- *IMPORTANT!* In DDC mode, the analogue outputs are generally muted because the unit's resources are used to perform a digital to digital conversion. They are not muted if the input is 4-wire DSD.
- *IMPORTANT!* Disconnect any cables from AES Reference In before using DDC mode. Reference In is directly connected to Reference Out.

2. Unit Settings Menu





2.1 Swap L/R Channels

Normally set to off. If you find the Left and Right outputs of your system are swapped over due to a connection error, set Swap to On to digitally swap them back. You should correct the error and set Swap to off. Available in PCM mode only.

2.2 Automatic Muting

Automatically mutes the analogue outputs if the selected AES/EBU data stream is flagged as Non Audio (e.g. from a CD-ROM). The **Mute** indicator will light and **Non Audio** appears on the display.

IMPORTANT! Automatic Muting gives useful protection for your ears and equipment against the full scale digital noise resulting from playing non-audio data. It should normally be set to On.

Set to Off to disable Automatic Muting.

2.3 Automatic Input Selection

When set to **On**, the unit will detect the inputs connected to an active source and select one in the priority order **AES 1**, **AES 2**, **AES 3**, **AES 4**, **BNC**. If the selected input is disconnected or turned off, the unit will select the input with the next highest priority.

You can override this and select a different input by pressing the appropriate button. Automatic selection takes over again when the selected input is disconnected or the source turned off.

2.4 Phase

Inverts the phase of one or both channels to correct a phasing error. Available in PCM mode only. The options are:

Normal Left Out of Phase Right Out of Phase Both Out of Phase

2.5 Dim Volume Setting

Sets the level of volume dimming, activated by one of the three user-defined **Function** buttons (see the next section). The options are 3, 6, 12 or 20dB. Available in PCM mode only.

2.6 Function Shortcuts

	Defines the function of the F1 , F2 and F3 buttons.
do this:	Navigate to this menu and choose Function 1. The display will change to:
	Function key 1 assign: * None 1
	F1 < F2 Clear F3 > ¹
do this:	Press F1 or F3 to scroll through the options or press F2 to reset to None. When you reach the option you want, press the OK button to accept it – it will
	be marked *.
do this:	You can set up Function 2 and Function 3 to different options as you like.
	The options are: None Dim Volume (PCM mode only) Sync Mode DDC Mode
	De-Emphasis (32, 44.1 or 48kS/s PCM only)
	Test Tone On (PCM mode only) Metering Speed Metering Type Peak Hold Setting Release Peak Hold Dim Display Lock Front Panel Swap Channels (PCM mode only) Input Mode

After timeout, pressing the **Function** button followed by the **F1**, **F2** or **F3** button will give quick access to the functions you have chosen, without having to access the menu. In most cases, press the button repeatedly to cycle through the available settings.

3. Display Menu



Figure 17 – Display menu

3.1 Brightness Settings

Sets the display brightness. Choose any setting from 01 to 15. Press **F2** while this menu is open to set the default brightness (10).

3.2 Display Timeout

Sets the time delay before the menu closes automatically and the display changes back to the normal display (status or meter display). The options are: 15 Seconds, 30 Seconds, 1 Minute, 3 Minutes or No Timeout. The last option disables the automatic timeout – use the **Cancel** button to return to the normal display.

3.3 Lock Front Panel

Protects the front panel control settings from unauthorised changes. Normally set to Unlocked. When set to Locked and a control is pressed, the warning **"Locked! Hold Cancel to Unlock"** appears at the bottom of the display.

3.4 Display Language

Sets the language of text appearing on the display.

Use the **F1** and **F3** buttons to scroll through the choices or **F2** to default to English. Press **OK** to accept the choice. In this release, English, French, Dutch and German are available.
4. Connector Settings Menu



Figure 18 – Connector settings menu

4.1 Sync Source

Sets the source to which the digital data is synchronised. There are two submenus:

4.1.1 Sync Mode

Choose a synchronisation source:

Audio Input	Slaves to the selected audio input (this is the normal setting).
Slave to	
Word Clock	Slaves to a Word Clock connected to the Clk In
	connector. Ensure the BNC Settings menu (see
_	page 38) is set to SDIF.
Slave to AES	
Reference	Slaves to an AES3 reference clock connected to
	the Reference In connector. Set the Reference
	Mode menu (see page 38) to suit.
Ensura the DDC M	ade many is set to off when using the Poference In

IMPORTANT! Ensure the DDC Mode menu is set to Off when using the Reference In connector.

4.1.2 Auto Slave

When set to Off, the unit slaves to the input set by the Sync Mode menu. When set to On, automatically slaves to **Reference In** if it is present, over-riding the Sync Mode setting.

4.2 AES Settings

This menu sets the behaviour of the AES interfaces. It has one sub-menu:

4.2.1 Reference Mode

Sets the behaviour of the AES reference loop. There are three options:

Loop Through Use this setting if the unit is in the middle of an AES reference daisy chain. The Reference In connector presents a high impedance to the reference source to avoid loading problems. Loop Terminated Use this setting if the unit is at the end of a reference daisy daisy-chain, or if it is the only unit connected to the reference source. The Reference In connector presents a 110Ω load to the reference source to correctly terminate it. Slave to Audio Input The Reference Out connector generates an AES3 clock slaved to the selected audio input. Do not connect to Reference In when in this mode. **IMPORTANT!** Ensure the DDC Mode menu is set to Off when using the **Reference In** connector.

4.3 BNC Settings

This menu sets the behaviour of the four BNC connectors in the **SDIF/DSD** interface. It has one sub-menu:

4.3.1 Set PCM Format

Sets the format of the **SDIF/DSD** interface. There are two options:

AES	Configures Clk In to accept AES3 encoded data at TTL levels at up to 96kS/s. Clk Out sends AES3 clock (no data) at the same rate as the data on Clk In .
SDIF	Configures the four BNC connectors as an SDIF interface for PCM or DSD operation. This is the usual setting.

4.4 Messaging

Displays the message flags encoded in the selected AES3 data stream. If the unit is not locked to an AES3 data stream, the display shows:

Messaging: No input available No message to display

If the unit is locked to an AES3 stream, the message can be displayed in one of two formats:

4.4.1 Simple View

The message flags are decoded and displayed. The display is updated every 6 seconds. Scroll down to see the message flags further down the list:

Messaging:

Professional Flag set or Consumer Flag set Non Audio Flag not set or set Emphasis: No Emphasis, 50/15µs, CCITT J17 or Not Indicated Destination: (null) Origin: (decoded from input) Sample Rate: 32, 44.1 or 48kS/s Multiplying factor: x1, x2, x4, x8, x0.125, x0.25, x0.5, x2/3, x3/7, x6/7 Mode: Two Channel, Single Channel, Primary/Secondary, Stereo or Not Indicated Reference Signal: Grade 1, Grade 2 or Not set

4.4.2 Full Details

Scroll through the message bytes, one after the other:

Messaging: Byte 00 = xxxx Byte 23 = xxxx

See the Appendix at the back of this manual for details of the message bytes.

4.5 Locking Bandwidth

For best jitter performance or when using a master clock, this menu should normally be set to Fine.

Some digital audio equipment (even some quite expensive products) produce data streams with jitter outside the AES3 specification. This can upset the DAC, causing intermittent data errors or muting. If this problem arises, setting the Locking Bandwidth to Coarse may solve the problem, at the expense of degraded sound quality. Only use Coarse lock where degraded audio is better than no audio! For more information on this topic, see "Jitter and PLL Bandwidths" on page **86**.

4.6 RS232 Address

Displays and allows access to the unit's RS-232 identity code (an address between 00 and 99). This is used by the Windows Remote Control software to send specific messages to specific units. Use the \uparrow and \downarrow buttons to change the setting of a digit. **F1** or **F3** selects the other digit and **F2** sets that digit to 0.

IMPORTANT! When using the Windows Remote Control, each *dCS* unit in the daisy chain MUST be set to a different RS232 address.

The RS-232 control formats and procedures are covered in more detail in the section "RS-232 Remote Control Interface" on page **77**.

5. Metering Menu



Figure 19 – Metering menu

5.1 Peak Hold

The peak hold feature adds a pair of dotted lines to the right of the bar meters. The peak hold lines track the largest peak registered. To turn peak hold on, select Always Hold. To turn off peak hold, select Never Hold.

The peak hold reading can be reset at any time, using the Release Peak Hold feature, available as one of the Function Shortcuts (see page **35**).

5.2 Metering Type

Sets the style of meter display. The options are:

Meter Both Displays both average and peak responding bargraph meters. The order of the bars is left peak (top), left average, right average, right peak (bottom).

50 40	30 es	- ejo -	- (S -	Ű.	÷ŝ.	1 (0	
						_	
44.	<u>1kS/s</u>				<u>24 -</u>	Bits	

Absolute

Values Displays the absolute signal levels as two numbers in dB0, left / right. This mode is intended to be used when calibrating with tone, not for monitoring music.

-37.3	-37.3
44.1k5/s	24 Bits

Bars

Displays peak responding bargraph meters, left at the top. The attack is instant, the decay is set by the Metering Speed menu. The response is that of a peak program level meter.



History

Shows a running history across the display of the highest signal level on either channel. The graduation dots at the right side are full scale (top), -6dB0, -12dB0 and -18dB0.



Average

Displays average responding bargraph meters, left at the top. The reading is an average of the samples collected in the last 300ms. The response is similar to a VU meter.

50.40 (30	න් න	(s	(i)	ŝ	
44.1kS7	's			24	Bits

5.3 Metering Speed

Sets the decay time for peak responding meters. The options are:

FastDecays by 50dB per secondSlowDecays by 10dB per second

With Meter Type set to History, the screen contains 1.6 seconds of data when set to Fast and 80 seconds of data when set to Slow.

5.4 Metering Boost

Normally set to No Boost.

If you are monitoring low level signals, change the setting to 20dB Boost to boost meter displays by 20dB, improving readability. Meter calibration is also shifted by 20dB. This affects all meter types.

6. Test Menu



6.3 Tone Frequency

Sets the frequency of the tone generator.

Set the frequency in the same way as for Tone Level. You can set frequencies between 1Hz and 99.999kHz, but the response will be limited by the analogue board anti-image filter and DC filter. The output is better than 0.1dB down from less than 10Hz up to 50kHz.

6.4 Test Display

A test routine for the LED indicators and main display.

- Each LED flashes dim red then bright red.
- Each LED flashes dim green then bright green.
- The mains display turns off, then full on 4 times.

6.5 Phase Check	
	Use this feature to check if one channel in your system is phase inverted.
do this:	Lock the unit to a digital source and press the Menu OK button to start the test.
	The unit will display In-Phase and output noise in phase on both channels. If there is no phase inversion, this should produce a central image. If there is a phasing error, it will not.
	After 6 seconds, the display will change to Out Of Phase and the noise on one channel will be inverted. If there is no phase inversion, this should not produce a central image. If there is a phasing error, it will produce a central image.
	The unit will continue flipping between these two states until you press the Cancel button.
6.6 Channel Che	ck
	Use this feature to check if the stereo outputs in your system are swapped.

do this: Lock the unit to a digital source and press the Menu OK button to start the test.

The unit will display **LChannel** and output modulated tone on the left channel only. After 6 seconds, the display will change to **RChannel** and the modulated tone will appear on the right channel only.

The unit will continue flipping between these two states until you press the **Cancel** button.

6.7 Simulation Test

This test allows you to simulate the effect of reducing the word length to less than 24 bits (for example, to record on a 16 bit recorder), add Noise Shaping and listen to the result. There are two sub-menus:

6.7.1 Noise Shaping

If the Output Word setting is less than 24 bits, always apply some Noise Shaping to smooth out the transitions. The options are:

off	Use for 24 bit words only.
1st to 9th order	Apply a higher order as the word length falls.
Auto	If in doubt, this option will apply a sensible Noise Shaping order.

6.7.2 Output Word

Select the required word length from 24 bits to 16 bits and ensure Noise Shaping is set to match.

IMPORTANT! Do not accidentally leave the unit set to less than 24 bits – you may be disappointed by the sound quality.

7. Information Menu





7.1 Software Revision

Displays the Front Panel and Main Board software versions.

7.2 dCS Contact Details

Displays *dCS*' address, phone numbers and e-mail address.

7.3 Unit Temperature

Displays the internal temperature, close to the crystal oscillators, in both Celsius and Fahrenheit.

7.4 Unit Identification

Displays the unit serial number – this includes important configuration information. Please quote this number (all of it) if you contact your distributor or dCS for support.

7.5 Distributor Details

Displays contact information for your *dCS* distributor.

7.6 CD Download

Allows software update from a *dCS* CD.

do this:	Connect a CD player or transport to an AES input or the SDIF-2 input and select that input to lock the <i>955</i> to it. Ensure the unit is in PCM mode, not DSD.
do this:	Load up a dCS CD containing updated 955 software and ensure the player is NOT playing.
do this:	Start the CD Download routine.
	The unit will display "Erasing Code", then "Please Start The CD".
do this:	Set the CD player to PLAY.
	The unit will display "Scanning CD" for about 10 seconds, then "Please Skip To Track n", where n is a number.
do this:	Advance the CD player to the required track.
	The unit will display " Downloading – Please Wait ", together with two progress bars. The lower bar fills rapidly then clears and adds a block to the top bar. When the top bar is full, the unit displays " Finished – Please Cycle Unit Power ".

do this: Disconnect power, either by pulling out the power cable or using the rear panel switch. Wait 10 seconds then reconnect power and switch on.

If the update includes new display software, the unit will display "New Front Panel Software – Press OK To Download It".

do this: Press the **OK** button and wait while the unit boots up.

What can go wrong with a CD Download?

If you try to run a CD Download with the unit unlocked or not locked to a suitable CD player, the unit displays "**Error Not Locked To A 44.1 Source**" and aborts the procedure.

If the unit finds that the CD is not the right *dCS* CD, is some other CD or is unreadable, it will display "**Invalid CD**" and abort the procedure.

If power is interrupted for any reason during a download, the software may be corrupted. If this happens, it is very likely that the Windows Remote Control will still work, allowing a dCS distributor to download new software from a PC. If you are very unlucky and the Remote Interface routines are corrupted, a dCS distributor can download new software from an EPROM.

8. Store Menu



Figure 22 – Store menu

8.1 Recall Settings

Recall settings from User Stores 1 to 6 or from Fixed Stores A to L. Some of the Fixed Stores may be empty in this release.

do this: Scroll to the store you want, press the **OK** button to select it, then press the **F** button stated on the display to confirm (or **Cancel** to reject it).

The unit will display "**Recalling – please wait**" while the settings are loaded. A progress bar will appear on the display for about 10 seconds, followed by "**Done**".

IMPORTANT! The unit will not load stores named "Empty" or with an erased name.

Fixed Stores

In this release, the Fixed Stores are loaded with the following set-ups:

Store A – Factory Reset

DAC	PCM format, Filter 2, Auto De-Emphasis, DDC Off.
Unit	Swap Off, Auto Mute On, Auto Input Select On,
	Phase Normal, Dim Volume 12dB.
Display	Bright 10, Timeout 15s, Front Panel Unlocked,
	Language unchanged.
Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref
	Loop Terminated, BNC to SDIF, Lock Bandwidth
	Fine, RS232 Address 05.
Meters	Always Hold, Meter Both, Fast, No Boost.
Test	Tone Off, Simulation Test – NS Off & 24 bits.

If any of these settings are not mentioned in the set-ups detailed below, they are the same as for Store A.

Store B – Multi-channel DSD

DAC	DSD format, AES1-4, Filter 1.
Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref
	Loop Through, BNC to SDIF.

do this: Connect up as shown on page **21**. Set the AES Settings menu for the first unit in the chain to Slave to Audio Input, for the last unit in the chain to Loop Terminated.

Chaining the AES Reference line ensures accurate alignment between DACs.

Store C – Multi-channel P3D with Master Clock

DAC	DSD P3D format, AES1-4, Filter 1.
Connector	Sync Mode - Audio Input, Auto Slave On, BNC to SDIF.

do this: Connect up as shown on page 22.

Slaving the DACs to a Master Clock such as a dCS 995 ensures accurate alignment with no special setups.

Store D – Multi-channel PCM

DAC	PCM format, AES1-2, Filter 2.
Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref
	Loop Through, BNC to SDIF.

do this: Connect up as shown on page 20. Set the AES Settings menu for the first unit in the chain to Slave to Audio Input, for the last unit in the chain to Loop Terminated.

Chaining the AES Reference line ensures accurate alignment between DACs.

Store E – P3D

AES Ref
Α

do this: Connect a P3D source to the AES 1 – 3 inputs.

Store F – DSD Quad

	Store G-DSD S	SDIF
do this:	See page 19.	
	DAC Connector	DSD format, AES1-4, Filter 1. Sync Mode - Audio Input, Auto Slave On, AES Ref Loop Terminated, BNC to SDIF.

	DAC	DSD format, BNC, Filter 1.
	Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref Loop Terminated, BNC to SDIF.
do this:	See page 18.	

Store H – P3D DDC

DAC	DSD P3D format, AES1-4, Filter 1, DDC Mode On.
Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref
	Loop Terminated, BNC to SDIF.

do this: See page 24. Take the DSD SDIF OUTPUT from the CH1, CH2 and Clk Out connectors.

Store I – DSD4 DDC

DAC	DSD format, AES1-4, Filter 1, DDC Mode On.
Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref
	Loop Terminated, BNC to SDIF.

do this: See page 24. Take the DSD SDIF OUTPUT from the CH1, CH2 and Clk Out connectors.

Store J – PCM DDC

DAC	PCM format, AES1-4, Filter 2, DDC Mode On.
Connector	Sync Mode - Audio Input, Auto Slave On, AES Ref
	Loop Terminated, BNC to SDIF.

do this: See page 24. Take the single AES output from the AES Ref Out connector. Do not connect anything to AES Ref In. You can manually select a Dual AES input, instead of Quad AES. Note that the Analogue Outputs are muted in this mode.

Store K - Empty

Store L – Empty

8.2 Save Settings

Stores the current unit settings in User Stores 1 to 6.

do this: Set up the unit as required, open the <u>Save Settings</u> menu and scroll to the store you want to save to. Press OK, then press the **F** button stated on the display to confirm (or <u>Cancel</u> to reject it).

You will be asked to edit the store name. Names can be up to 16 characters long and can contain upper and lower case letters, numbers, punctuation marks, etc.

- do this: Scroll up and down to change the first character, then press **F2** to move to the next character. To insert a character, press **F1**. To delete a character press **F3**.
- do this: When you have finished editing, press **OK** to accept (or **Cancel** to reject).

You can rename an existing store:

do this: Open the <u>Save Settings</u> menu and scroll to the store you want to rename. Press the **F3** button and edit the store name as described above. Press **OK** to accept (or **Cancel** to reject the change). You can erase an existing store name:

do this: Open the Save Settings menu and scroll to the store name you want to delete. Press F1, then press the F button stated on the display to confirm (or Cancel to abort). Wait a few seconds while the store is erased.
do this: You must rename this store before it can be recalled.

THE HARDWARE – CONTROLS AND CONNECTORS

Front Panel



Figure 23 – Front Panel

Key to Front Panel

Α	Power On/Off button
В	AES 1, AES 2, AES 3, AES 4 and BNC input
	selector buttons with indicators
С	F1, F2 & F3 function short cut buttons
D	Display button
E	Mute button with Mute indicator
F	Main Display
G	Menu Operation buttons
Н	Volume Up / Down (1 / J) buttons
I	Function button

Note that most of the buttons have multiple functions.

On/Off Button

This button doubles as a power on / off switch and a standby mode switch.

IMPORTANT! Ensure the power switch on the rear panel beside the mains inlet is On.

do this: To switch on, press the **On/Off** button briefly. If power is available, the *955* will run through the power up routine.

Note that the **On/Off** button will not click when turning power on – this is normal.

do this: When you have finished using the unit, press the **Function** button, followed by the **On/Off** button, then the **OK** button to set the unit to standby mode.

The Analogue Outputs will mute, all displays will turn off. In this mode, the *955* uses very little power.

- do this: To restore normal operation, press the **On/Off** button briefly again.
- do this: To switch off completely, hold down the **Function** button and press the **On/Off** button.

Input Selection

The AES 1, AES 2, AES 3, AES 4 and BNC buttons are used to select inputs and formats.

The following sequence assumes that the Input Format menu setting is PCM and Automatic Input Selection is Off.

- do this: Press the AES 1 button once to select the AES 1 input. Press it again to select AES 1 + 2 Dual wire. Press it a third time to select AES 1-4 Quad wire.
- do this: Press the AES 2 button to select the AES 2 input.

do this: Press the AES 3 button to select the AES 3 input. Press it again to select AES 3 + 4 Dual wire.

do this: Press the AES 4 button to select the AES 4 input.

do this: Press the BNC button to select the SDIF/DSD input.

Alternatively, the active input may be selected from the Input Selection menu (see page **31**).

If the Input Format menu is set to DSD, pressing any **AES** button selects **AES Packed DSD** (DSD packed into AES 1 - 4) while pressing the **BNC** button selects **BNC Packed DSD** (DSD packed into 3 BNC connectors).

If the Input Format menu is set to P3D, pressing any AES button selects **AES Packed DSD** (DSD packed into AES 1 - 3) while pressing the **BNC** button selects **BNC Packed DSD** (DSD packed into 3 BNC connectors).

The input indicators over the five buttons glow red if the corresponding input is active but not selected. They turn green when the corresponding input is selected.

F1, F2 and F3 Buttons

These three buttons are user-defined by the Function Shortcuts menu (see page **35**) to allow quick access to several menu features. Pressing the **Function** button displays the settings of the three buttons, for example:

50 40	3jo	eis ein	(s	(a	. ș	::	0
-1S9n	cMd	F20	leEm	P	F3I	Fil	: ter

While the three functions are displayed at the bottom of the display, press the appropriate \boxed{F} button repeatedly to cycle through the options for that feature. Stop when you reach the required setting.

Display Button

Press the **Function** button and then the **Display** button to toggle between Status display and Meter display.

Mute Button

do this:

If the unit is locked to a digital source, press the **Mute** button to mute or unmute the Analogue Outputs. The **Mute** indicator will light when the outputs are muted.

If the unit is not locked to a digital source or the data is flagged as Non-Audio (for example a CD-ROM), the Analogue Outputs are automatically muted, the **Mute** indicator is lit and the **Mute** button has no effect. See Automatic Muting on page **34** if you need to over-ride an incorrectly set Non-Audio flag on a disk or tape.

Main Display

The main display tells you what the 955 is doing.

• The Status display shows the lock status (**Unlocked** or sample rate), the selected filter, the selected input and the sync source.



• The Meter display shows the selected level meters, the lock status (Unlocked or sample rate) and either the selected input (when unlocked) or the word length.



• When adjusting the Volume setting (PCM mode only), the bottom row of the display changes to **Vol** -**xx.x.** It changes back to the normal display 6 seconds after the last change.



- When accessing the menu (see page 26), menu pages are displayed here.
- If an error occurs during power up, the details of the fault will be displayed. See "Error Messages" on page **98**.

Menu Operation Buttons

This group of four buttons (\uparrow , \downarrow , **Menu OK** and **Cancel**) give access to the menu. See "Using the Menu" on page **27** for details.

(The two LEDs above this group are for a future enhancement.)

Volume Up and Down (\uparrow / \downarrow) Buttons

When in PCM mode with the menu closed, these two buttons set the digital Volume control between 0.0dB and -120.0dB in 0.1dB steps. All analogue outputs plus the DDC Mode digital output are affected. The Volume control is not available in DSD modes.

Function Button

Press the **Function** button to activate the **Display** button, **On/Off** button and the user-defined function buttons **F1**, **F2**, **F3**. You must define the functions of the three **F** buttons before using them for the first time - see the **Function** Shortcuts menu on page **35**.

Rear Panel



Figure 24 – Rear Panel

Key to Rear Panel

J	Balanced Analogue Outputs CH1(L) & CH2(R) on XLR male connectors
K	Sensitivity trimmers for Balanced Outputs only
L	Unbalanced Analogue Outputs CH1(L) & CH2(R) on RCA phono connectors
Μ	AES/EBU Reference In & Reference Out on XLR female / male connectors
Ν	AES/EBU digital inputs AES 1, 2, 3 & 4 on XLR female connectors
0	SDIF / DSD digital inputs CH1 & CH2 on BNC connectors
Р	SDIF Wordclock In & Out
Q	Remote In & Out – RS232 interface with loop output, 9-way male 'D' type connectors
R	IEC power inlet
S	Mains fuse holder
Т	Power switch

Balanced Analogue Outputs

If your pre or power amplifier has balanced inputs, connect the *955*'s **Balanced Analogue Outputs** to them. For best results, connection of these outputs to unbalanced inputs is not recommended – use the **Unbalanced Analogue Outputs** instead. Use good quality, screened, XLR audio cables.

Sensitivity trimmers

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

Unbalanced Analogue Outputs

For unbalanced equipment, use the **Unbalanced Outputs** with good quality, screened, RCA phono audio cables. The **Unbalanced Outputs** are very high quality! It is much better to use these for equipment with a single-ended input, than to use half of a balanced output. Doing so may cause ground loop hum pickup.

Reference In / Out

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. If several units are to be daisy chained with the same reference source, a terminating resistor may be turned off using the AES Settings / Reference Mode menu (see page **38**).

When in PCM mode with the DDC Mode menu set to On (see page 33), the sample rate converted digital output appears on **Reference Out**.

IMPORTANT! Disconnect any cable from **Reference In** before using **Reference Out** as a data output.

Use with 110 ohm screened, twisted pair cable designed for digital audio or RF. Do not use unscreened cables as they may pick up interference from other appliances.

AES 1, AES 2, AES 3 & AES 4 Digital Inputs

Four AES/EBU inputs which may be used independently or in groups of two (Dual AES on AES 1 & 2 or on AES 3 & 4), three (P3D on AES 1, 2 & 3) or four (Quad AES or 4-wire DSD on AES 1, 2, 3 & 4).

IMPORTANT! When using multi-wire formats, please ensure the AES inputs are correctly connected to the source, otherwise the channels will be swapped or mono'ed. This usually involves connecting AES 1 (or AES A) on the source to AES 1 on the 955, AES 2 (or AES B) on the source to AES 2, and so on.

Use with 110 ohm screened, twisted pair cable designed for digital audio or RF. Do not use unscreened cables as they may pick up interference from other appliances.

SDIF/DSD CH1, CH2 Data Inputs

In normal operation they are inputs for SDIF-2 encoded PCM, or SDIF-2 or SDIF-3 encoded DSD. The unit will auto-detect the data format. They are both TTL level signals for a 75 ohm line. They can be set to accept TTL level AES3 coded signals, using the BNC Settings / PCM Format menu (see page **38**). When used as an input, a suitable Wordclock generated by the source must be connected to **Clk In**.

In DDC mode, they can be used as data outputs, for re-formatting DSD data (see page **33**).

Clk In / Clk Out

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 page **65** for the time alignment of these signals.

Additionally, they can both be set for TTL level AES3 coded signals, using the BNC Settings / PCM Format menu, see page **38**. In this mode, **Clk Out** carries an AES3 clock, locked to the currently selected single AES input. The **Clk In** signal may be just an AES clock, for locking purposes, or a full AES3 coded input, selected by the **BNC** button.

Remote In & Out

If the WindowsTM 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 (see page 19). In addition, the unit may be software upgraded without removing the lid by downloading new software via the **Remote In** port – see page 94.

Connect up Remote ports using a 9-way screened cable, fitted with 9-way female '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 unit as com port to first unit. Suitable cables are available from dCS.

IMPORTANT! While the Windows Remote Control is connected and active, the front panel controls are disabled.

Mains Supply

Fused and filtered IEC (CEE22) mains connector. May also be switched on some units.

Additional Information

As well as connectors, the rear panel displays the following information about the unit:

Mains VoltageThe actual voltage setting supplied.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 **955 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.

dCS 955 TECHNICAL INFORMATION

Converter Type

• 5-bit, approximately 3MS/s oversampled, *dCS* patented *Ring DAC* topology.

Digital Interface Specifications

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

Table 1 – AES/EBU Interface Electrical Characteristics

SDIF & Word Clock	Input	Output	
Туре	Single end refe	ed, ground rred	
Impedance	100	25	Ω
Sensitivity (unloaded)	TTL	TTL	
Connectors	BNC	BNC	

Table 2 – SDIF / Word Clock Interface Electrical Characteristics

Sample Rates

- 32, 44.1, 48, 88.2 or 96kS/s on single wire interfaces and SDIF-2 / Word Clock interface, auto selected.
- 88.2, 96, 176.4 or 192kS/s on **Dual AES** interface, auto selected.
- 176.4 or 192kS/s on Quad AES interface, auto selected.
- 2.822MS/s for DSD, requires a 44.1kHz word clock.

Frequency Response (set to Filter 1)

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- Fs=32kS/s +/-0.1dB, 10Hz to 15kHz
 - Fs=44.1 or 48kS/s +/-0.1dB, 10Hz to 20kHz
 - Fs=88.2 or 96kS/s +/-0.1dB, 10Hz to 35kHz
 - Fs=176.4 or 192kS/s +0.25/-0.35dB, 10Hz to 55kHz³
 - DSD +0.2/-0.35dB, 10Hz to 55kHz



Figure 25 - Frequency responses at all sample rates, set to Filter 1

Volume Control

- Digitally implemented, 0 to -120dB in 0.1dB steps.
- Not available in DSD modes.

De-Emphasis

- Auto-selected, 50/15µs, CCITT J17 or none.
- Available in PCM mode at 32, 44.1 or 48kS/s only.

³ The point of high sample rates is not to have a ruler flat response with a brick-wall filter. The roll-off is deliberate. See for example "Effects in High Sample Rate Audio Material" by M.J.Story, R.J.Kelly, D.A.Macleod & M.N.Harris, presented at the 20th Tonmeister Tagung at Karlsruhe in 1998.

Analogue Output Specifications

	Balanced	Unbalanced	
Туре	Balanced,	Single ended,	
	semi-floating	ground referred	
Source impedance	< 3	52	Ω
(20Hz-20kHz)			
Maximum load	600	600	Ω
Noise, unweighted	< -110	< -110	dB0
(20Hz-20kHz)			
Spurious responses	< -100	< -100	dB0
(20Hz-20kHz)			
Signal balance @ 1kHz	> 40	-	dB
50Hz	45	-	dB, typical
1kHz	53	-	dB, typical
20kHz	53	-	dB, typical
L-R crosstalk,	< -100	< -100	dB
(20Hz-20kHz)			
Level for full scale	+14	+8	dBu
(off load)	3.88	1.95	V rms
Trim range	+/- 6	-	dB
Connectors	XLR male (2)	RCA Phono (2)	
Connections	Screen on pin 1	-	
	+Signal on pin 2		
	-Signal on pin 3		

Table 3 – Analogue Output Electrical Characteristics

Clocking

The sample clock quality significantly determines the output performance of the converter. The highest quality clocks that are available are crystals, so we use these. The *955* uses two on-board voltage controlled crystal oscillators (VCXO's) as clock sources - one for the 44.1kS/s related sample rates and one for the 48kS/s related sample rates.

Only one VCXO is active at a time. In slave mode, the active VCXO is synchronised to the clock signal extracted from the input by a phase locked loop (PLL). This PLL is of a special narrow bandwidth type, that provides a significant degree of "clock cleaning". The PLL is also 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.

Synchronising to source

Pull-in range± 300 ppm about nominal frequencyLock-in time< 6 seconds for most situations</td>

Power requirements

Units may be set for 100, 115/120, 200, 215/220 or 230/240V (+/-10%), 50/60Hz AC operation.

	Typical power consumption	Maximum power consumption
dCS 902 / 904	26W	30W
dCS 905	31W	36W
dCS 952 / 954	28W	34W
dCS 955	33W	39W
dCS 972 / 974	16W	20W
dCS 992	15W	18W
dCS 995	21W	28W

Table 4 – Power consumption for professional products

Size and Weight

	Length	Depth	Height	Weight
1U Units	430mm	395mm	44mm	6.8kg
	16.9"	15.6"	1.75"	15.0lbs
2U Units	430mm	395mm	88mm	8.5kg
	16.9"	15.6"	3.5"	18.7lbs

Table 5 – Size and weight for professional products

- All models are 1U high except *dCS* 972 and *dCS* 974 which are 2U.
- Removable rack ears take the overall width to 483mm (19").
- Removable feet add an extra 8mm (0.32") of height.
- The control knob on *dCS 972* and *dCS 974* protrudes an extra 22mm (0.87") from the front panel.
- Allow extra depth for cable connectors.

Operating Conditions

The *955* 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 Heat or Unit Temperature menu on page 44). 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 non-condensing.

.Figure 26 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 (1.2") between units gives reasonable cooling





Figure 26 – Temperature rise above ambient for a unit in a stack of 3 with poor ventilation

Group Delay

The group delay for a *dCS* 904 ADC and *dCS* 954 DAC⁴ 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 6 – *dCS* 904 ADC group delay in μ s, 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 7 - dCS 954 DAC group delay in µs, 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.

⁴ The other *dCS* Pro ADCs and DACs use the same DSP code modules and the same digital engine, so these results apply.

Anti-Image Filtering

The *955* offers a choice of 4 anti-image filters on most sample rates. These filters affect the ultrasonic part of the spectrum - 20kHz 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 images⁵ 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 1ms). 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 PCM mode filters that we have included give increasingly good energy smear performance, and consequently have increasingly relaxed roll off. Filter 1 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, Filter 2 is preferred while Filters 3 & 4 are popular with users recording rock and jazz. Opinions differ widely – so try them for yourself.

The *955* 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 ripple⁶, depending on the filter being used. The stop band is typically below – 110dB0 and can be as low as –130dB0.

⁵ See, for example "Principles of Digital Audio", 3rd Edition, by Ken C Pohlmann (McGraw-Hill Inc, 1995)

⁶ Filters always have some ripple. For "zero ripple" filters this is in the μ dB to pdB region.

Sample Alignment

The *955* aligns samples such that Word Clock Out aligns with AES3 samples out^7 , 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.









⁷ In this section, AES Out is the AES Reference Out set to slave to the audio input.

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 29 - Word Clock In to Word Clock Out, 96 kS/s



Figure 30 – 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 40ns. Input is at the top of the displays, output is at the bottom. Signals are at the sockets on the *955*, the unit was slaved to AES input.



Figure 31 – AES3 in to AES3 out, 96kS/s



Figure 32 - AES3 in to AES3 out, 44.1kS/s





Figure 33 – Word Clock In to AES3 Out, 96kS/s



Figure 34 - Word Clock In to AES3 Out, 44.1kS/s

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 / Ref Mode setting. In Loop / Loop Through or Loop.t / Loop Terminated modes, it is copied from the Reference In data. When set to Route / Slave to Audio Input, the message is copied from the lowest numbered AES input selected. When set to ddc / DDC Mode On 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 footnote⁸, 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 AES3 Go?

The AES/EBU format was designed to go reasonable distances, at 44.1kS/s and 48kS/s. **Figure 35** and **Figure 36** below show it over 16 m and 94 m (the source is the top trace) 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 96kS/s (twice the data rate the format was designed for) the allowed cable length is less. **Figure 37** and **Figure 38** 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 96kS/s cable runs to 20 m or less, and using good cable near this length.







Figure 38 – AES3 format at 96kS/s over 94 metres

DSD

DSD (Direct Stream Digital) is a single bit very high sample rate (2.822MS/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.



DSD Format, showing Full Scale Signal dCS 972 SW v1.54

Figure 39 – DSD, showing DSD full scale

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

Four DSD formats are available:

DSD SDIF-2 DSD SDIF-3	2 data channels plus a Word Clock (see page 75). 2 data channels with embedded clock (see page 75).
DSD Quad	2 channels packed into 4 AES3 links for storage on
P3D ⁹	2 channels packed into 3 AES3 links for more compact storage on a 6-track 24-bit 44.1kS/s recorder (see page 76).

⁹ The P3D interface may be optional on some models.
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 message 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 (Word Clock (top) and two data channels) at 44.1 kS/s and 96 kS/s.



Figure 40 – SDIF-2 PCM format at 96 kS/s

hp running



Figure 41 – 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 (15μs, 50μs)	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 8 – 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.









DSD on SDIF-3

SDIF-3 embeds a clock in the SDIF-2 data stream. The *955* does not extract the embedded clock and so still requires a word clock to decode SDIF-3. SDIF-3 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 – the software looks for 150ms – 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 **Loop Out** port of one unit and the **Remote In** port of the next unit. The same type of cable is used to connect the **Remote In** port of the first unit on the chain to the COM port of a PC (see page **19**).

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 seconds 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", page **78**.

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 50ms 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:

- хх
- 00 immediate (less than 50ms)
- 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 50ms 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 ³/₄ 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 100ms. Resets ALL units on daisy chain to 1200 baud.

GLOBAL ADDRESSES

Address F0 hex (240 decimal)

ALL units on the daisy chain react to the 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 the daisy chain react to the 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.

1) 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

2) 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

3) 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

3) 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

4) 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 5) 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	ommand yte	Number of arameters in Command	Parameters	'arameters in Response	2	Ŋ	D D	SIK .
	с m	~ ~ ~		<u> </u>	A D	Dم	DD	MO
RS_AUTO_SLAVE	15	1	0 = do not automatically slave 1 = automatically slave to a reference input	0	X			Х
RS_MASTERSLAVE	16	2	First parameter 1 = Master 0 = Slave. If slave, second parameter: 0 = AES 2 = SDIF	0	Х		X	X
RS_ENABLE_DEBUG	19	3	Global Command	None	Х	Х	Х	Х
RS_SEL_FS	32	1	Select Output Fs	Echos message	Х	Х	Х	
RS_FILTER	33	1	Select Filter, 0-3	0	Х	Х	Х	
RS_EMPH	34	1	Select De-emphasis filter to use, 0 = Auto 1 = 50/15 2 = CCITT 3 = None	0	X	Х	Х	
RS_OUT_MODE	36	1	0 = Output SDIF wordclock on w/clk out, 1 = Output AES on w/clk out	0	Х	Х		Х
RS_TRUNC	39	1	No. of output bits (16 - 24)	0	Х		Х	Х
RS_SNS	40	1	Noise shaper, 0 = Auto 1 = Off 2 = 1st Order 3 = 3rd order 4 = 9nth order	0	X		х	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	Х	Х		Х
RS_OUT_RATE	42	1	0 = Low speed output (e.g. Dual 88.2/ Quad 192 1 = High speed option	0	Х		Х	Х
RS_MUTE	43	1	0 = Unmute 1 = Mute	0	Х	Х	Х	Х
RS_AUTO	44	1	1 = Turn off automatic input selection (DAC)	0		Х		
RS_7SEGS	47	1	1 = Turn off 7-segment display	0	Х	Х		Х
RS_INP_FORMAT	48	1	Select Input format for DACs – 0 = Auto 1 = Single wire 2 = Dual Wire 4 = Quad	0		Х	Х	
RS_4WIRE	49	1	0 = Enable 4-wire DSD outputs 1 = Disable 4-wire DSD outputs	0	Х			
RS_FLIP	50	1	0 = Normal 1 = Flip channels	0		Х	Х	

Command name			Parameters	(0				
	Command Byte	Number of Parameters in Command		Parameters in Response	ADC	DAC	DDC	MCIK
RS ACUT	51	1	1 = Disable Auto Digital Muting	0		Х		
RS_FINE_LOCK_MODE	52	1	1 = Use coarse lock 0 = use fine lock	0		Х		
RS_WAVETYPE	63	1	0 = Signal Generator Off 1 = Signal Generator ON	0	Х	Х	Х	Х
RS AMP	64	1	Generator Amplitude, format X	0	Х	Х	Х	Х
RS_FREQ	65	1	Generator Frequency. Specified as a 32 bit number. Expressed as a fraction of Sample Frequency	0	х	х	х	Х
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	Х	Х		х
RS_OVLD_LEV	87	1	Overload threshold, format X	0	Х			
RS_VOL	111	1	Digital volume control, format X	0		Х	Х	
RS_PHASE	112	1	Phase: 0 = None inverted 1 = Both inverted 2 = left inverted 3 = right inverted	0		Х	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	Х	Х	Х	
RS_BAUD_RATE	141	1	Global command	None	Х	Х	Х	Х
REQUEST_DSD_MODE	142	0	response -> DSD mode	Yes				
REQUEST_FREQUENCY	143	0	response -> Frequency of unit	Yes				

Table 9 – 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.

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 nanoseconds, 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 as a delay in the signal path.

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

Using your dCS 955 for the first time

Thank you for purchasing this *955*. Before attempting to use your *955*, please read at least the rest of this section and the "Step-by-Step Guide" on page **10**. This will enable you to set the unit up quickly with your system.

What's in the Box?

The box should contain the following:

- dCS 955
- User Manual
- Quick Start Guide or Menu Guide
- Power Cable
- Spare Fuses (2)
- Remote Interface Cable
- Windows Remote Control Software

Check the contents of the inner carton very carefully against the list above. Notify your re-seller as soon as possible if anything is missing or damaged. dCS suggest that all of the original packaging is retained for use when transporting any units. Replacement packaging can be ordered from dCS or our distributors.

Your re-seller should arrange for you to receive a dCS **Extended Warranty Certificate** within 30 days of purchasing a new dCS 955. If it does not arrive, please contact dCS. Please keep the **Extended Warranty Certificate** in a safe place. For details of the warranty, see page 93.

Safety Notice

The dCS 955 contains no user serviceable parts. Do not attempt to open the case as there are potentially dangerous voltages present inside. In the event of the unit developing a fault, please consult your distributor.

IMPORTANT! This equipment MUST be connected to a safety earth (or ground) via the power cable.

Mains Voltage Setting

Before connecting the power cable to your *955* for the first time, please check that it has been set to the correct operating voltage for your local mains supply. 50Hz or 60Hz operation is not important – the unit can use either. The unit's present voltage setting is shown on the label beneath the mains inlet on the rear panel. If this does not match your local supply voltage, DO NOT attempt to use the unit. Contact your re-seller to arrange to have the unit reset. Using the *955* with the wrong mains setting for your local supply may result in serious damage to the unit and will invalidate the warranty. DO NOT attempt to reset the unit yourself.

Positioning the Unit

Place the *955* on a firm, vibration free base so as to allow convenient connection to your digital source. Keep it away from sources of heat, such as power amplifiers, radiators or direct sunlight.

Installing the Unit in a Rack

The unit is supplied with 19" rack mount ears fitted. If it is rack mounted, use the ears to locate the unit in the rack and stop it sliding forward – but:

IMPORTANT! The ears are not strong enough to be used as the only mechanical support.

A rack-slide option is available for this purpose, see page **90**.

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

OPTIONS

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

Option code	Option
VID	Video sample rates
RS	Rack slide
V5	Mains voltage set to 230/240V
V4	Mains voltage set to 215/220V
V3	Mains voltage set to 200V
V2	Mains voltage set to 115/120V
V1	Mains voltage set to 100V

Table 10 – Options available

Locking to Video Sample Rates

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

Rack Slide

All *955* units will accept a pair of telescopic rack slide rails to facilitate 19" rack mounting.

Mains Supply Voltage

Any unit may be set for operation from 230/240V, 215/220V, 200V, 115/120V or 100V A.C. Units are shipped set for the mains supply voltage according to the destination. The voltage setting can be updated later by your distributor, if necessary. Specify the new country of use or the new voltage setting.

Having Your Options Changed

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

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

MAINTENANCE AND SUPPORT

Service & Maintenance

dCS audio products are designed not to need regular maintenance, and contain no user serviceable parts apart from the mains fuse.

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

Mains Fuse

There is a mains fuse near the power inlet, accessible from the outside of the unit. If the fuse blows, it may be changed by the user. The current consumption of the unit is very low, so it only blows if power surges occur or there is a fault in the unit. Usually, power surges cause no other damage, 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 T HRC fuse

IMPORTANT! If the fuse should fail, it is essential that it be replaced with one of the same type and rating. Failure to do so could result in damage to the unit, risk of fire or electric shock and will invalidate the guarantee.

Replacing a Blown Fuse

Referring to **Figure 44**, remove the power cable, use a small flat bladed screwdriver to pry up the tab on the fuse carrier (A) and pull it out. Push the blown fuse out of the clip in the carrier (B) and dispose of it. Fit a new fuse in the clip (C) and push the carrier back into the unit so that it clicks home. Spare fuses are provided with the unit.



Figure 44 – Changing the Mains Fuse

IMPORTANT!

Disconnect the power cable before changing the fuse.

Warranty

Initial Warranty

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

Extended Warranty

The period of the Warranty cover may be extended to 12 months from receipt by the first owner at no extra cost, by registering your purchase with dCS. Your distributor should have arranged this on your behalf by filling in an **Owner Registration** form at the time of sale and returning it to dCS. If you have bought a new dCS 955 and do not receive an **Extended Warranty Certificate** for this unit within 30 days of purchase, please note the serial number on the underside of the unit and contact dCS.

If you sell the unit within the warranty period, the balance of the Extended Warranty may be transferred to the new owner by completing the Owner Registration Transfer form (see page **113**), and returning it, with the **Extended Warranty Certificate**, to dCS.

Warranty Exclusions

The Warranty on this *dCS* 955 shall be void if:

- the product is misused in any way.
- any unauthorised modifications or repairs are carried out.
- the product is not used in accordance with the Operating Conditions stated on page 62 of this manual.
- the product is serviced or repaired except by an authorised *dCS* Distributor.
- the product is operated without a mains earth (or ground) connection.
- the unit is returned inadequately packed .

Obtaining Service

If you experience problems with your dCS 955, you should check the "Troubleshooting" section on page 98. You may be able to resolve the situation yourself (for example, by changing a menu setting).

If this does not resolve the problem, contact your authorised dCS distributor for advice, quoting the model, the full serial number (see page 44), the software version number (see page 44) and giving a detailed description of the fault. If further action is necessary, your distributor will advise you fully. When returning a unit, the original packaging should be used to avoid transit damage. Replacement packaging sets may be purchased from dCS.

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

Updating your *dCS* 955

dCS products make extensive use of software configurable chips – FPGAs and DSPs. This gives us the ability to update our products to add extra features, update digital interface standards or make performance improvements by loading new software. Occasionally, a hardware update may be necessary also to increase the "capacity" of the electronics, add extra connectors or extra front panel controls.

Software Updates

Please note that not all software updates make an earth-shattering change. You should have a clear idea of what you expect to gain before updating to the latest issue.

If the software loaded in your unit is (for example) version 3.45:

- A change to version 3.46 indicates a minor update for internal *dCS* use to make testing easier, or more thorough, or to cater for some minor hardware change. Do not update your unit.
- A change to version 3.50 is a more serious update, offering extra functionality. If you want to use the extra features, update your unit. The manual will be updated for this (check the web-site).
- A change to version 4.00 is a major update. It will require updated hardware and the manual will be updated

If you have older hardware, some of the features added by new software may not be available due to (for example) a missing connector. For **recent** hardware, this is mentioned in the "Upgrader's notes" in this manual. If you need the missing feature, contact your re-seller / distributor or dCS to arrange a hardware update.

- If the software currently loaded includes a CD Update or CD Download page, you can update the software yourself. Contact *dCS* for a free CD-ROM and follow the instruction in the CD Update / CD Download section on page 44. The update program will check your hardware configuration and install the latest software compatible with it.
- If you do not have the CD Update / CD Download feature loaded or have difficulty with it, you can download files from a PC into any unit fitted with a BOOT ROM (units manufactured from 1999 onwards) as described in the next section. *dCS* supply the files free, contact us for details.
- For older units without a BOOT ROM, your distributor can install new firmware in your unit. Firmware updates are low-cost from *dCS*. Contact your distributor for details.

Installing New Software from a PC

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

Using the RS-232 download is hands free, but takes about 40 minutes 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.

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 (<u>www.dcsltd.co.uk</u>). 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 DAC will display **Prog**. The Windows programme will display **Erasing Flash** (10 seconds), 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 minutes or so). After this has counted down, the PC displays **Done** and the DAC 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 display **Hold** on its front panel. Do not do anything at this stage. When that message disappears, it will be ready for normal use.

Hardware Updates

You may wish to have your hardware updated from time to time to take advantage of new features in the latest software. *dCS* offer this service - we will retest, reset any adjustable items to current shipping standards, and install any modifications or updates that have occurred since your unit was first shipped.

The price will depend on the hardware changes necessary – please contact your dealer or dCS for details and pricing. In order to ensure speedy turn around, please contact us prior to returning the unit to get a **Service Return** number.

Safety and Electrical Safety

There are no user serviceable parts inside the *955* and so there is no need to remove the covers. If for some reason you do:

IMPORTANT! Disconnect the power cable before removing any covers or changing the fuse.

There are no substances hazardous to health inside the 955.

Cleaning the Case

Do not apply any of the following cleaning products to the case as they will damage or alter the finish:

- Corrosive or abrasive agents
- Spirit or alcohol based cleaners
- Wax polish
- do this: To remove dust, wipe with a moist, soft cloth.
- do this: To remove deposits from the case, first disconnect the power cable then spray very lightly with a proprietary glass cleaner containing ammonia and wipe off gently with a soft cloth, taking care to avoid scratches. Do not spray the cleaner onto the connectors or the rear panel.

TROUBLESHOOTING

Error Messages

If a fault is detected during the power-up sequence, the unit will display one of the following messages:

- FPGA's Not Loaded
- FPGA's Broken
- DSP's Broken

This can be caused by a power line transient during power-up. Try disconnecting the power cable for 10 seconds, then powering up again.

If the fault persists and the power supply / connection is stable, the hardware may have been damaged. Please contact your distributor to arrange repair.

Troubleshooting Your System

If you experience difficulties when using your *dCS 955*, the following suggestions may help to resolve the problem.

The unit fails to power up

- Ensure there is power available on the power cable and the rear panel power switch (if fitted) is on. Press the **On/Off** button.
- 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 on page **92**.
- Check that the power cable is pushed fully home into the power inlet in the rear of the unit.
- If there are signs of activity on the front panel but the unit does not boot up, try disconnecting power and powering up again. This can be caused by power line transients.

The unit fails to lock to a source

- Ensure the correct input is selected.
- Check that the digital audio cable is of the correct type, correctly connected and not damaged. Damaged or unsuitable cables are a VERY common cause of malfunctions!
- Check that the source is switched on.
- Some types of source equipment do not generate a digital output when idling set the source to "play" mode and check that the *955* locks.
- Try setting the Sync Source / Sync Mode menu to Audio Input.
- If you are syncing to AES Reference In, try setting the Reference Mode menu to Loop Terminated and disconnect any cables fitted to Reference Out.
- If you are using SDIF-2, set the BNC Settings / PCM Format menu to SDIF.

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 or off occupies the microcontroller for about 15 seconds.
- Short mains supply drop-outs may occasionally 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 distributor.
- The Windows Remote Control program disables the front panel controls while it is running.

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 the unit is locked to the source you want to monitor.
- Ensure Mute is not enabled LED off.
- Set the Volume control to 0.0dB. Ensure Dim Volume is not active.
- Ensure the source is sending audio data.
- If the DDC Mode menu is set to On, change it to Off. With PCM inputs, DDC Mode disables the analogue outputs.

The unit remains muted, Mute button has no effect

- If the unit is unlocked, the display will indicate this.
- The source data may have the Non-Audio flag set. If the Automatic Muting menu is set to On, the unit will remain muted. If you are SURE the source data is playable audio, turn the system volume down and set the Automatic Muting menu to Off.
- If using P3D format, check that the three wires are connected correctly.

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

- 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.
- Ensure the trim tool is small enough to fit the slot in the level trimmers.

The Left and Right channels are swapped

- Check that the Swap L/R Channels menu is set to 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 or AES A) 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 or AES B). See the manual of the source equipment for information.
- In Quad AES or DSD Quad 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 source equipment for information.
- In **SDIF PCM** 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 PCM or DSD modes, ensure both CH1 and CH2 inputs as well as Clk In are connected correctly.

The output is monophonic

- If the unit is locked to one wire of **Dual AES** or **Quad AES**, one signal will appear on both outputs. 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.
- Do not use unscreened cables.
- Try setting the Locking Bandwidth to Coarse. If this solves the problem, the source equipment is likely to have a high level of jitter you should consider changing it.
- This can be caused by slaving some, but not all, of the system components to a Master Clock. If the unit is being driven by an upsampler or DDC, you will probably need a Master Clock that can output different sample rates, such as the *dCS* 995 or 992. Otherwise, slave the 955 to the upsampler / DDC outputs.

The output is full scale noise

- The source may be non-audio data such as a CD-ROM.
- If you are using **DSD Quad**, check that the cables are connected correctly.
- Check that the Input Format menu is correctly set to match the digital data.

The unit fails to slave to a Master Clock

- If you are using an AES Reference, connect this to Reference In and set the Reference Mode menu item to Loop Through or Loop Terminated as appropriate. Set Auto Slave to On. Ensure DDC Mode is off.
- If you are using Word Clock, connect this to Clk In. Set the BNC Settings / PCM Format menu to SDIF and the Sync Mode menu to Slave to Word Clock.
- 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.

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 source is AES and the sample rate is 48kS/s or lower, the De-Emphasis setting may be incorrect. Set the De-Emphasis menu to Auto.
- 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). Set the De-Emphasis menu to 50 / 15us, CCITT J17 and None in turn until the correct setting is found. When you change to a different disk or tape, we recommend setting the De-Emphasis menu to Auto again.

The sound quality is poor

- Ensure the Simulation Test menu is set to Noise Shaping Off and Output Word Length 24 bits.
- If using an Upsampler or DDC, ensure its' Output Word Length is set to 24 bits.
- If the Locking Bandwidth menu is set to Coarse, try setting it to Fine. Only use the Coarse setting if you experience problems locking to your source equipment – it does not give best results unless the source is very jittery.
- If you are using a DSD format, your power amplifier may be reacting badly to the high level of ultrasonic noise. Try setting the Filter Settings menu to Filter 2 or Filter 3.
- If you are using a DSD format, ensure the Filter Settings menu is not set to Filter 4 this is intended for metering only.

The display is very dim

• The Brightness Setting in the Display Menu is set to a low level. Open the menu and change to a high setting.

If this information does not resolve the problem, please contact your distributor for advice.

IF YOU NEED MORE HELP

In the first instance, you should contact your dealer / re-seller or distributor. If he cannot resolve the issue, contact *dCS*. Our office hours are 8:30 a.m. to 5:00 p.m. Monday to Friday, UK time (UTC or UTC + 1hr). Contact us by phone or fax on:

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

Table 11 – *dCS* Contact Information

You can write to us at:

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

Our email address: more@dcsltd.co.uk

Our web-site is: www.dcsltd.co.uk

The web-site is regularly updated. You will find full details of all dCS products here, plus the latest dCS news.

Other Information

dCS produce technical notes from time to time, on issues related to ADCs, DACs and DDCs. If you are interested in these, please do not hesitate to contact us.

INDEXES AND SOFTWARE VERSION NUMBERS

Software History

This manual is for *dCS* 955 software version 1.0x.

Definitions and Abbreviations

dB	A relative signal level or ratio in decibels. The context may indicate the reference level.
dBu	A signal level relative to 0.775V rms, making no
	allowance for external loading.
dB0	Level in decibels, referred to a full scale sine wave
	in a sampled system. So, 0dB0 is full scale.
kS/s	Sample rate in kilo-samples per second. This replaces kHz, which is technically incorrect when
	referring to sample rates.
DAC	Digital to Analogue Converter, sometimes referred
ADC	Analogue to Digital Converter comptimes referred
ADC	Analogue to Digital Converter, sometimes referred
	to as an A/D Converter.
DDC	Digital to Digital Converter, sometimes referred to
	as a D/D Converter.

Key to Cable Identification

Cable Type	Colour / Style	
XLR Analogue	Brown, solid	
RCA Phono Analogue	Brown, long dash	·
XLR Digital (AES3)	Blue, solid	
BNC Digital	Pale blue, solid	
RCA Phono Digital (SPDIF)	Pale blue, short dash	
Optical (SPDIF)	Dark magenta, solid	
IEEE 1394	Red, solid	
Sync Link	Green, solid	
Wordclock	Dark green, solid	
AES Reference	Dark green, long dash	·
GPS Reference	Dark green, short dash	
RS-232	Purple, solid	

If you are reading a colour print or a soft copy of this manual, cable types shown in figures can be identified from **Table 12**.

Table 12 – Cable colours and styles

Where more than one cable of any type is used in a drawing, they can be identified by a coloured sleeve at the source and a coloured arrow head at the destination.

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Selling your *dCS* 955 Privately

If you dispose of your 955 by a private sale within the warranty period, you will enhance its value if you transfer the balance of the Extended Warranty to the new owner. Please fill in the Owner Registration Transfer document on the next page, sign it and return it to dCS, together with your Extended Warranty Certificate within 30 days of the sale.

On receipt, we will send a new Extended Warranty Certificate to the new owner.

Owner Registration Transfer

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

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

Serial No:

(located on the underside of the unit)

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

Name of registered owner: Signed:

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