

DSA-1

Operation Manual

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A file named `READ.ME' may be included with the software; it includes information relevant to this manual. The purpose of the `READ.ME' file is to provide the most up-to-date information about program development; as such the file may post-date the latest manual issue or revision.

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1. Introduction

The DSA-1 is a hand-held instrument for measuring connectivity parameters of stereo digital audio signals in one of the formats described in AES3-1992 and IEC958 (commonly referred to as AES/EBU or SPDIF). The recently-proposed AES3-ID coaxial format is also supported. Testing of 110 Ω balanced, 75 Ω coaxial or TOSLINK optical media is possible. The DSA-1 can also generate a wide range of test signals, and can edit signals 'in-line'. The unit may be powered from its internal rechargeable battery or from an external DC power supply.

The parameters measured and signals generated by the DSA-1 are intended to provide a means of assessing 'connectivity'; i.e. the likelihood of trouble-free interconnection.

There are many factors which have implications for connectivity, for example the sampling-rate of the sending equipment or the number of bits present in the audio data word. The level and correctness of the Channel Status implementation is also important if proper connectivity is to be achieved.

The consequences of failure to meet connectivity requirements can range from complete failure, as when a source operates at a sample-rate not supported by a receiver, or the most subtle effects which result from truncation of the audio word, as when a 20-bit source is connected to a 16-bit device without provision for truncation processing such as re-dithering or noise-shaping.

Among the most frustrating of connectivity problems are those which lead to intermittent operation, where infrequent clicks or drop-outs mar audio quality. These may occur at random, or may be related to program content. Such problems are most often the result of inadequate operating margin in some electrical or timing parameter of the interface, due to either equipment or cabling problems.

The DSA-1 is a tool designed to measure connectivity parameters with great accuracy and simplicity. Its automatic Test Sequences can detect subtle abnormalities in interface conditions without the need for detailed analysis by the operator.

Although the DSA-1 includes a number of basic audio-data generation and analysis functions, it does not provide distortion measurements or other sophisticated audio analyses; the Prism Sound Dscope is a high-performance digital audio test system (including generator and analyzer) which focuses on signal content, rather than connectivity.

1.1. Summary of capabilities

The following section briefly summarises the capabilities of the DSA-1 version 2.0 firmware for the benefit of new operators. For operators experienced in the use of the DSA-1 version 1.0 firmware, new features are marked with an asterisk.

Carrier Analysis:

LED panel: Summarises major carrier faults (including biphasic or parity errors, poor eye-width, incorrect Channel Status block length and analogue content) and displays sampling rate and grade.

Sampling rate: Measures sampling rate (fs);
Measures sampling rate accuracy in ppm;
fs measurements may be calibrated to house reference;

Jitter: Measures fs Jitter (from source);
Measures Data Jitter (from cable);
Measures eye-pattern degradation;
De-jittered output available;
Selectable jitter corner-frequency;

Amplitude: Measures differential amplitude;
Measures common-mode component;
Measures audio-band component;

Phase: Measures carrier phase with respect to reference;

Data Analysis:

LED panel: Summarises audio, Validity, User and Channel Status bits;

Audio amplitude: * Bargraph meter with numerical readout in dBFS or hex;

Bit activity: Displays which bits are active in the audio word;

Channel Status: Displays binary data and functional description;

Test Sequences: Allow the operator to test incoming signal to professional or consumer standards at the press of a button, without detailed knowledge of the interface specifications.

* Operator or in-house standards authority may compose custom Test Sequences on a host PC which are then loaded into the DSA-1.

* **Generator:** Sine, square, triangle, ramp, pulse, Jtest, or noise functions;
Variable amplitude, frequency and audio word-length;
Fully programmable Channel Status, field by field;
Independent A and B channel settings;
Channel Status pattern can be 'cloned';
Programmable Valid bit;
Jitter generation;
Synchronises to AES3/11, wordclock or local reference;

Editing mode: In-line editing of Channel Status, field by field;
In-line editing of Valid bit;
In-line superimposing of generator audio signal;
In-line jitter modulation;

* **Channel Check:** Checks cables, tape machines, data links etc. by comparing every incoming audio bit within the selected word-length with the pseudo-random data generated at the output. Errors are logged against time and can be printed. Check is delay-independent, and source and checker can be remote from each other.

* **Watchdog:** Operator can identify CARRIER and DATA panel parameters to be 'watched', e.g. sampling rate accuracy, biphase errors, Validity bit, Channel Status CRC errors. Variations of the identified parameters are logged against time and can be printed. Distinct alarm strings for each watched parameter can be transmitted from the RS-232 port if required.

* **Microscope:** Digital gain up to 90dB can be applied to incoming audio, with a selected input word-length imposed, prior to the monitoring DAC and DO outputs. This allows evaluation of dithering and noise-shaping schemes.

*** Results Log:** Measurement results, settings etc. can be printed immediately or held in non-volatile memory for later printing or uploading to a host PC.

Monitoring DAC: Feeds headphones or integral loudspeaker;

Test Inputs: XLR (AES3), coaxial (AES3-ID, SPDIF), optical (CP340);
Copper inputs have switchable termination for in-line use;

Test Outputs: XLR (AES3), coaxial (AES3-ID, SPDIF);

Ref Sync Inputs: XLR (AES3/11), coaxial (AES3/11, wordclock);

'Scope Outputs: Isolated, unbalanced, x10, 75 Ω signal plus programmable sync;

1.2. System options

The DSA-1 digital audio interface analyzer comprises:

- The DSA-1 instrument
- A pair of phono-to-BNC adapters for SPDIF measurements
- The DC power adapter, plus local IEC mains lead
- A null-modem serial cable for host PC (Prism part # DSA1/NMC)
- This manual
- The Host Support software on 3.5" diskette

The following optional accessories are available:

- Serial printer cable (Prism part # DSA1/SPC)
- Deluxe carrying case for all of the above (Prism part # DSA1/CC)

DSA-1 Host Support software (for IBM-compatibles) comprises the following items:

- Results Log Uploader/Editor program (for MS Windows)
- Test Sequence Compiler/Downloader program (for MS Windows)
- Macro Processor program (for DOS)
- Firmware Loader program (for DOS)
- Latest-issue DSA-1 firmware (program in DSA-1's FLASH memory)
- READ.ME software information update (where applicable)

Updates to the DSA-1 firmware and Host Support software will be supplied on 3.5" diskette (contact Prism Media Products or your distributor for 5.25" media).

Full details of how to use the DSA-1 Host Support software suite are given in section 7.

EMC STATEMENT

The DSA-1 and its DC power supply conform to all applicable norms and standards for EMC prevailing in the countries in which it is sold.

However, the DSA-1 is a precision instrument capable of measuring carrier artifacts (such as timing jitter) at low levels. Operation of the DSA-1 in the presence of strong electromagnetic fields may result in a reduction in accuracy of some measurement modes.

1.3. Using this manual

This Operation Manual is provided as a guide to the complete DSA-1 system. The manual is divided into sections as follows:

- Introduction and 'getting started'
- Display, control and menu references
- Informal guide to measurement techniques
- Specification
- Appendices and index

In this manual various conventions are used to identify keys, LEDS etc. on the DSA-1, and also on the host PC keyboard:

A DSA-1 key name (or legend) is enclosed in square brackets thus []; for example the ON key on the front panel of the DSA-1 would be referred to as [ON].

When references are made to the contents of the LCD display of the DSA-1, the display contents are shown in a rectangular box, for example:

Prism Sound
DSA-1 sw2.0 hwC

Where LED indicators or legends on the DSA-1 are referred to in the text, single quotation marks are used, thus:

'CHARGE'

would refer to the yellow LED above the [ON] key.

In examples of command line or other dialogue with a host PC, similar conventions are used:

Triangular brackets refer to keys on the host PC, thus where you see <Enter>, the reference is to the large Enter key on the keyboard of your PC, usually also labelled with an arrow symbol thus '↵'. Single keys such as 'G' or 'B' are indicated as <G> or .

Where references are made to command-line entries on a host PC, these use a Courier font, like this:

```
C:> DIR<Enter>
```

2. Getting started with the DSA-1

First, unpack your DSA-1 and check that you have all the items listed in the inventory in section 1.2. Check that the packaging and its contents are undamaged; if not contact your distributor or Prism Media Products Limited immediately.

The DSA-1 may be operated from its internal rechargeable battery or from the external DC power adapter supplied. The type of DC adapter may vary according to the country in which the product is sold, in order to achieve compliance with local regulations concerning equipment safety. Check that the DC adapter supplied is suitable for use with your mains voltage.

The DSA-1 may be supplied from the factory with the battery uncharged, in which case the DC adapter must be used in order to operate the brand-new unit. You may either operate from the DC supply and charge the battery later, or charge the battery immediately as described below.

2.1. Equipment overview

The layout of the DSA-1 controls and connections is designed for maximum ease of use in both 'hand-held' and 'desk-top' situations.

In the centre of the front panel are located 'CARRIER' and 'DATA' display panels, wherein a number of LED indicators normally provide constant monitoring of the basic properties of the signal-under-test.

Below these panels are a row of four keys for control of the monitoring D/A converter (DAC) and the red **[ON]** key. At the bottom of the front panel are a row of I/O selection controls which affect the operating modes of the DSA-1's connectors.

The upper part of the front-panel contains an LCD display and its associated keys which provide a menu system for making detailed measurements, running Test Sequences, and controlling all additional functions of the DSA-1.

The main test I/O connectors are situated along the right-hand edge of the DSA-1, and the auxiliary-function connectors are along the front edge. All these connectors are legended around the edge of the front panel. The headphone socket is at the near right-hand corner of the DSA-1.

2.2. Turning on the DSA-1

The DC input connector is on the front right-hand corner of the instrument under the front-panel legend 'DC IN'. Connect the DC adapter to the mains, and plug its output into the unit's DC input connector.

Switch on the DSA-1 by pressing the **[ON]** key. If all is well the '**CHARGE**' indicator will light, followed shortly afterwards by the '**ON**' indicator.

The display will contain the message:

Prism Sound
DSA-1 sw2.0 hwC

This message indicates that version 2.0 of the DSA-1 firmware is installed, and that the DSA-1 hardware is at revision C. If a later version of firmware has been installed, at the factory or elsewhere, the current version will be displayed instead of V2.0 as above. The READ.ME file on the system software diskette will also contain information about the current firmware version and revision level.

If battery operation is required, then the battery may first need to be charged; this will take approximately 12 hours in fast-charge mode. The battery charge mode must first be selected and this is described in section 6.9.5. While operating on external DC power the unit will trickle-charge the battery.

If no test input is connected (and if the Generator is off), all indicators in the DATA display panel will be unlit and in the CARRIER panel the '**FAIL**' indicator will be lit.

If the behaviour of your DSA-1 is not as described above then contact Prism Media Products or your distributor.

2.3. Connecting a digital audio input

Connect a digital audio source to the appropriate DSA-1 input connector. If possible, use a source which allows some control of the audio output. Ideally, use a digital audio test generator, such as the Prism Sound Dscope system, which will allow sufficient degree of control to demonstrate most of the features of the DSA-1. Alternatively, basic operation can be demonstrated using a simple CD player (with a digital output), or a DAT recorder.

If you wish to connect a consumer source via an RCA (phono) cable, you will need to plug in one of the phono-to-BNC adapters (provided with the DSA-1) to the co-axial 'DI' input.

Having connected a suitable source, select the required input format (optical, co-axial or balanced) using the **[DI]** selector on the lower left of the DSA-1 control panel. The **'FAIL'** LED on the CARRIER panel should extinguish and a sampling frequency and accuracy should be indicated.

The DATA panel shows the basic activity of the audio and status data; if a 16-bit source has been chosen only the main **'AUDIO'** indicator should be lit, whilst a 24-bit source would cause both **'AUDIO'** LEDs plus the **'AUX'** LED to be lit. Turning the source off may cause the **'AUDIO'** and **'AUX'** LEDs to be extinguished, unless a DC offset or dither signal remains (as is often the case with CD sources).

If you are using a Prism Sound Dscope system as a test source, try incrementing the generator word-length from its minimum setting (12-bit) to its maximum (24-bit) and observe the bit-activity indicators. If you are using a CD or DAT player, the **'AUX'** indicator and the left-hand **'AUDIO'** indicator should be off, representing a 16-bit source.

The **'V'** indicator is normally unlit unless the source is 'invalid' (such as when a CD player is stopped or paused). The **'U'** LED is lit if there is activity in the User channel. The lower part of the DATA panel contains a brief display of the meaning of the incoming Channel Status bits.

2.4. Some basic tests

This section assumes that the settings of the DSA-1 are the factory defaults. If you have difficulty in reproducing the results below, please try again after instating the default settings via menu 9.3.

Press the **[LS]** key so that its **'ON'** indicator is lit, and set the **[L.S. MODE]** to **'BOTH'**. A mono mix of the two audio channels should now be audible from the DSA-1's loudspeaker. It may be necessary to operate the **[^]** and **[v]** keys to obtain the correct volume level.

Press the **[EXIT]** key a few times to ensure that you are at the top level of the menu system, and then use the **[<]** and **[>]** keys until the display shows:

```
Select Mode:
1. Carrier Tests
```

Now press the **[ENTER/NEXT]** key, and the display will show:

```
1. Carrier Tests
1.1. fs Freq
```

On pressing the **[ENTER/NEXT]** key again, the measured sampling rate of the test signal is displayed, along with its accuracy in ppm (parts per million), for example:

```
fs = 44100.44 Hz
Acc = +10      ppm
```

Now press the **[EXIT]** key followed by the **[>]** key. The display shows:

```
1. Carrier Tests
1.2. fs Jitter
```

Enter the sampling-rate jitter test mode by pressing **[ENTER/NEXT]**, and the result is displayed, for example:

```
fs Jitter [P-P]
= 4.3ns fc=1k5
```

These few basic steps are intended to give an introduction to the workings of the DSA-1. Later sections of this manual cover all DSA-1 operations in detail.

2.5. Running an automatic Test Sequence

This section assumes that the settings of the DSA-1 are the factory defaults. If you have difficulty in reproducing the results below, please try again after instating the default settings via menu 9.3.

Press the **[EXIT]** key a few times to ensure that you are at the top level of the menu system, and then use the **[<]** and **[>]** keys until the display shows:

```
Select Mode:  
3. Test Sequence
```

Now press the **[ENTER/NEXT]** key followed by the **[>]** key, and the display shows:

```
3. Test Sequence  
3.2. Select
```

Press the **[ENTER/NEXT]** key again, and use the **[>]** key to select the Test Sequence which is appropriate to your test signal. If the DATA panel shows **'PRO'** select:

```
Selected: Sys03  
General Pro Test
```

Or if the DATA panel shows **'CONS'** select:

```
Selected: Sys02  
Consumer Test
```

Now press the **[EXIT]** key, followed by the **[<]** key to bring you to the 'Run' menu option. Press the **[ENTER/NEXT]** key to run your selected Test Sequence. You should see a series of messages describing the various parts of the test as they happen. If all is well, the LCD should eventually show:

```
General Pro Test  
* Tests complete
```

Or, if you were running the Consumer Sequence:

```
Consumer Test  
* Tests complete
```

On the other hand, you may discover that something is wrong with your test signal, in which case an error message is displayed instead. This may seem surprising but, in

fact, it is not very unusual since many digital audio signals currently in use do not completely conform.

Section 6.3. includes more details about selecting and running Test Sequences.

2.6. Try the Signal Generator

This section assumes that the settings of the DSA-1 are the factory defaults. If you have difficulty in reproducing the results below, please try again after instating the default settings via menu 9.3.

First, switch on the Signal Generator by pressing the **[DO]** key twice to light the **'FNC'** LED. The **'Fs'** section of the CARRIER panel should now indicate the sampling rate as **'48.0'** with no 'ppm' indicators lit: In the **'FNC'** mode, the **'Fs'** LEDs indicate the basic sampling rate of the Generator and not the test input.

Connect the XLR or coaxial test output ('DO') of the DSA-1 to a device capable of monitoring a digital audio input. You can use the DSA-1 for this by connecting one of the 'DO' outputs to the corresponding 'DI' input using an XLR or coaxial cable. In this case, make sure that the loudspeaker is turned on (by pressing the **[LS]** key to illuminate its **'ON'** indicator) and that the appropriate data input is selected (with the **[DI]** key in the lower left-hand corner). When this is accomplished, the **'FAIL'** LED will go out and the Signal Generator's 1kHz tone will be heard on the loudspeaker.

Press the **[EXIT]** key a few times to return to the top level of the menu system, and then the **[>]** key a number of times until you see:

```
Select Mode:
4. Generator
```

Press **[ENTER/NEXT]** and then **[>]** key a number of times until you get to:

```
4. Generator
4.4. Audio Param
```

Pressing **[ENTER/NEXT]** again will display the Audio Parameters menu:

```
→A&B:ON    -14dB
Wrd:24b    1000Hz
```

Operating the **[<]** or **[>]** keys adjusts the indicated parameter, in this case turning the generator on and off. The **'A&B'** indication means that adjustments are currently set to operate on both output channels.

Pressing the **[ENTER/NEXT]** key twice moves the flashing arrow onto the generator amplitude parameter, initially set at -14dBFS. Operating the **[<]** or **[>]** keys now reduces or increases the amplitude setting. The generator mutes while the setting is being adjusted, then turns on at the new amplitude. The **[EDIT/CLEAR]** key is used to return to default settings where appropriate, in this case re-setting an amplitude of -14dBFS. This amplitude is often used as a standard line-up level, being +4dB above -18dBFS.

Pressing the **[ENTER/NEXT]** key again moves the flashing arrow onto the generator frequency parameter, initially set at 1000Hz. Operating the **[<]** or **[>]** keys now adjusts the frequency in a similar way, with **[EDIT/CLEAR]** re-setting to the default 1000Hz if required.

This is a very brief introduction to the Signal Generator, which has a wide capability. The full details of the Signal Generator are explained in section 6.4.

3. DSA-1 I/O connectors

Around the periphery of the membrane panel on the front and right-hand edges are several grey panels indicating the function of the connectors located immediately below.

The following connectors are provided:

- DI (digital audio inputs) for measurement:
 - 110 Ω balanced
 - 75 Ω coaxial
 - OPTical TOSLINK
- DO (digital audio outputs) for loop-through, re-clocking and Generator modes:
 - 110 Ω balanced
 - 75 Ω coaxial
- REF SYNC inputs (AES11/IEC958 format or 'Wordclock'):
 - 110 Ω balanced input
 - 75 Ω coaxial input
- Headphone socket
- TEST-OUT connections for oscilloscope:
 - LINE test output
 - SYNC test output
- RS-232 serial port
- DC IN connector

3.1. DI (Digital-audio Input) connectors

DI connectors are provided for the signal-under-test in balanced 110 Ω XLR, coaxial 75 Ω BNC and optical TOSLINK formats. The desired connector is selected using the **[DI]** key in the lower left corner of the panel.

Any of the DI connectors may be driven with consumer or professional formats. The BNC connector will accommodate a phono (RCA) input for SPDIF measurements using the phono-to-BNC adapter provided.

The 75 Ω and 110 Ω terminations (at the BNC and XLR inputs respectively) are switched out if the **[TERM]** key selects '**HI-Z**'. This is useful for 'loop-through' testing wherein the DSA-1 can monitor a signal 'in-line'. If the **[DO]** control is set to '**RPT**' or '**FNC**', then the DI termination is over-ridden to '**LO-Z**' since 'loop-through' operation is not required.

The maximum input voltage is 10V P-P at the BNC connector, and 20V P-P at the XLR connector (which also has a maximum common-mode input of 20V P-P). **These voltages should not be exceeded, otherwise damage to the DSA-1 may result.**

Note that carrier amplitude measurements are not available when the optical input is selected, although all timing-related carrier measurements (such as jitter) are available.

3.2. DO (Digital-audio Output) connectors

The DSA-1 can pass the signals from its BNC and XLR DI inputs directly to a pair of respective BNC and XLR outputs. This allows 'in-line' measurements to be made. Alternatively, the DSA-1 can perform jitter suppression (or filtering) and signal re-clocking at these outputs (by selecting the '**RPT**' option with the **[DO]** key). In this case, the DO connectors source the regenerated signal. The DSA-1's Signal Generator function is available at these outputs by selecting '**FNC**'

The BNC DO connector is intended to drive a 75 Ω load, and the XLR DO connector a 110 Ω load.

When re-clocking (jitter filtering) or generating (i.e. when NOT in loop-through mode), the output signal amplitude at the BNC DO connector may be selected as described in section 6.9.1.2. The 'HIGH' amplitude voltage is 1V P-P for compatibility with the newly-proposed BNC version of the interface; the 'LOW' amplitude voltage is 500mV P-P for compatibility with consumer SPDIF systems. The output of the XLR DO connector in this mode is nominally 4.4V P-P. Note that these amplitudes assume a correctly terminated line.

3.3. REF SYNC input connectors

Two REF SYNC inputs are provided for provision of an fs reference signal. These are used primarily as external references for the Signal Generator, but are also used for calibration of the DSA-1's fs frequency measurements to an external rate ('Ref Learn' mode) and for measuring the phase relation between a signal-under-test and a house reference ('Phase v. Ref' mode).

The BNC REF SYNC input is terminated with 75Ω , and the XLR REF SYNC input with 110Ω . Either can be driven with consumer or professional signals according to IEC 958/AES11. The BNC input can alternatively be driven with a TTL-level sampling-rate clock of 50% duty cycle (i.e. a 'Wordclock' or 'SDIF2 Wordsync'). The DSA-1 will automatically determine which connector and which format is in use whenever an external reference is required.

The maximum input voltage is 10V P-P at the BNC connector, and 20V P-P at the XLR connector (which also has a maximum common-mode input of 20V P-P). **These voltages should not be exceeded, otherwise damage to the DSA-1 may result.**

3.4. Headphone socket

The headphone output is at the rear side of the unit below the **[SYNC O/P]** key. The connector is a standard 1/4-inch stereo jack socket and will drive high- or low-impedance headphones. The lowest impedance that should be connected is 8Ω . The left channel is connected to the tip, the right to the ring, and the outer connection is a common ground.

3.5. Test-out connectors

Two test outputs are provided to assist in the use of an oscilloscope for detailed examination of the signal-under-test.

An unbalanced version of the signal at the selected DI connector (including the optical connector) is routed to the LINE test-out connector, thus providing a convenient means of viewing the received signal without unbalancing, misterminating or otherwise upsetting the line. For best bandwidth and accuracy, the LINE test output is designed to drive a 75Ω termination at the oscilloscope and, under this condition, an attenuation of 10x results similarly to the case of a high-frequency oscilloscope probe. In many instances the termination may be omitted in which case a 5x attenuation occurs at the output.

The SYNC test-output provides a signal designed to be used as a trigger for the oscilloscope. It drives a pulse of arbitrary amplitude and duration, whose rising edge is timed just prior to the start of the selected preamble of the signal-under-test (or the Signal Generator reference if it is switched on). The **[SYNC O/P]** key is used to select whether the SYNC test-output is derived from the X, Y or Z preamble of the signal-under-test, or from the REF SYNC input. For best results, the SYNC test-output should also be terminated with 75Ω at the oscilloscope. Note that various interlocks operate on the **[SYNC O/P]** selection, for example accessing the 'Ref Learn' or Jitter measurement menus or using the Signal Generator may preclude the use of certain settings.

NOTE: Both the LINE and SYNC test outputs are electrically isolated from the DI and measurement circuitry so that results are not compromised by an unwanted ground connection to the oscilloscope.

3.6. RS-232 serial port

The DSA-1's serial port is generally used for communication with a PC-compatible host computer. At present, this port has the following functions:

- Uploading the DSA-1 Results Log to the Results Log Editor software;
- Downloading User-defined Test Sequences from the Test Sequence Compiler software;
- Printing the Results Log directly to a serial printer;
- Remote-control of the DSA-1 using the Macro Processor software or Serial Slave mode;
- Downloading firmware updates to the DSA-1;
- Issuing Alarm codes in Channel Check and Watchdog modes;

Details of these operations are covered in section 7.

The port employs a 9-way connector with the same pinouts as a PC-AT compatible computer, and for the majority of the above functions, a null-modem cable is connected between the serial ports of the DSA-1 and the host PC. Such a cable is available as accessory DSA-1/NMC from Prism Media Products, or from your distributor. Alternatively a null-modem cable can be assembled using two female 9-way 'D' connectors wired with a length of two-core-plus-screen cable as follows:

CN1 (9F)		CN2 (9F)
2	—————	3
3	—————	2
5	— [screen] —	5

For connection to a most types of serial printer, a cable is available as accessory DSA-1/SPC from Prism Media Products or from your distributor. Alternatively, a printer cable can be assembled using a female 9-way 'D' connector and a male 25-way 'D' connector wired with a length of two-core-plus-screen cable as follows:

CN1 (9F)		CN2 (25M)
2	—————	2
3	—————	3
5	— [screen] —	7

NOTE: The RS-232 interface connections are electrically isolated from the DI and measurement circuitry so that results are not compromised by an unwanted ground connection to the PC.

3.7. DC IN connector

The 'DC IN' connector is used for battery charging and remote power, and is a standard 2.1mm DC 'barrel' connector with the centre terminal positive. A supply voltage of 9V to 12V at a maximum current of 1A is required. In order to ensure safe and compliant operation, **it is strongly advised that the DSA-1 only be used with the DC adapter supplied.**

4. Indicator panels

In the central part of the DSA-1 front panel are two display panels which continuously display certain basic information about the signal-under-test. Apart from a few special cases, these panels are always active.

The left-hand panel is called the 'CARRIER' panel, and usually displays only information about the incoming carrier waveform. The 'DATA' panel on the right displays information about the data content of the signal-under-test.

4.1. CARRIER panel indications

The LEDs in the upper row of the CARRIER panel indicate serious carrier defects, whilst the lower two rows indicate the sampling-rate of the signal-under-test (or the Signal Generator, if operating) and the accuracy thereof.

4.1.1. FAIL indicator

This is lit when no signal is detected at the selected DI input, or when the signal is of such a poor quality that the DSA-1 cannot decode it properly. The DSA-1 receiver is very tolerant, and therefore any signal which cannot be decoded by the DSA-1 is unlikely to be recognised by other equipment.

When the '**FAIL**' LED is lit, all other indicators on the CARRIER and DATA panels are off, since they would have no meaning in this state (except in '**FNC**' mode, when the 'Fs' LEDs reflect the sampling rate of the Signal Generator).

4.1.2. ERROR indicator

The '**ERROR**' indicator lights to show that a biphase error has been detected; it remains lit for a short time to make momentary failures more easily visible.

The interface standard dictates that transitions of the carrier always occur at certain times, and never at certain other times. The presence or absence of extra transitions embodies the varying binary data. A biphase error occurs when a transition of the incoming carrier occurs where none is allowed, or when an expected transition is missed. This is explained further in appendix A.

A sub-frame parity error is a type of biphase error, but a biphase error does not necessarily imply a parity error.

4.1.3. N.FAIL indicator

This LED is lit when a poor quality carrier is present, but is being successfully decoded. Specifically it shows that the 'eye-width' of the carrier is reduced below 50% of the ideal value. This would render the signal irrecoverable to most receiving equipment.

4.1.4. BLOCK/PHASE error indicator

This LED has a dual function: When the DSA-1's Signal Generator is turned off, the **'BLOCK'** LED indicates that an incorrect number of samples has been counted in between successive 'Z' preambles of the signal-under-test.

The Z preambles define the start of the Channel Status block (sometimes also the User data block). The interface standard requires that Z preambles are normally 192 samples apart. A BLOCK error can cause Channel Status to be wrongly interpreted in some equipment.

When the DSA-1's Signal Generator is switched on, this LED indicates a **'PHASE'** error, specifically that the phase of the incoming carrier differs from the Generator's reference by more than the $\pm 25\%$ ($\pm 90^\circ$) which the interface standard states that a receiver must tolerate.

This mode of the indicator allows an asynchronous input to be recognised; in the case of a small frequency error, the LED 'beats' at the frequency difference. Note that measurement of incoming jitter or eye parameters is not possible when a PHASE error is flagged.

4.1.5. A'LOG indicator

The **'A'LOG'** (analogue) LED indicates the presence of a substantial signal below 20kHz at the selected measurement input. This indication normally suggests accidental connection of an analogue source, or perhaps a miswired interconnection causing mains 'hum'. The threshold is nominally -25dBu (46mV rms).

4.1.6. fs frequency display

The middle row of LEDs on the CARRIER panel indicates the source sampling frequency (fs). If the source frequency deviates by more than 1000ppm from one of the standard frequencies, the **'OTHER'** indication lights. The fs frequency LEDs are labelled in kilohertz (kHz).

The sampling frequency LEDs change function when the DSA-1 Signal Generator is operating. In this state, the '**Fs**' LEDs indicate the basic sampling rate of the Signal Generator, rather than the test input signal.

4.1.7. fs accuracy display

The fs accuracy indicators at the bottom of the CARRIER panel provide a guide to the accuracy of the source sampling frequency, in units of ppm (parts-per-million). If none of these LEDs is lit, then the source fs is within 1000ppm of the standard rate, unless the '**OTHER**' LED is lit. A more precise measurement of sampling frequency accuracy is available through the menu system.

The right-hand LED is designated '**WRT REF**'; if this LED is lit, it indicates that the DSA-1 has been software-calibrated to an external Ref Sync as described in section 6.9.4. In this case, the accuracy measurement is made assuming that the external Ref Sync was absolute.

The '**ppm**' LEDs are extinguished when the DSA-1 Signal Generator is operating; in this state, the '**Fs**' LEDs are indicating the basic sampling rate of the Signal Generator, rather than the test input signal.

4.2. DATA panel indications

The 'DATA' panel is situated in the right-centre of the DSA-1 front panel. Its function is to provide a quick indication of the activity of the data bits in the sub-frame, i.e. the audio and the various ancillary bits. Note that the contents of the DATA panel apply to only one sub-frame (channel) at a time, as selected by the **[CHAN]** key at the bottom of the unit.

4.2.1. Sub-frame display

The upper row of DATA panel indicators shows a facsimile of the audio and status bits of the incoming signal-under-test in transmission order, i.e. how they might appear if the sub-frame were to be displayed on an oscilloscope.

The **'AUX'** LED shows non-zero activity in the auxiliary sample bits (these would normally be the four LSBs of a 24-bit audio sample, or they might be used for another purpose such as provision of a lower-quality, secondary channel). The left-most **'AUDIO'** LED represents audio bits 4-7 of the 24-bit field, which could be the four LSBs of a 20-bit sample. The right-hand **'AUDIO'** LED represents the sixteen most-significant bits of the sample.

In the general case, the **'AUX'** and **'AUDIO'** LEDs can be regarded collectively as an indicator of active word-length, of which the right-most three, two or one LEDs may be illuminated to signify a 21–24-bit, a 17–20-bit or an up-to-16-bit word-length respectively.

The **'V'** LED shows the state of the Valid bit, and is ON when the audio data is 'INVALID'. The exact meaning of the Valid bit has not been rigidly or consistently defined in the interface standards. The 'INVALID' condition is most often described as when the audio sample is 'unsuitable for conversion', for example when non-audio data is carried; however, the idea of using the Valid bit to flag interpolated or imperfectly corrected audio samples has often been mooted. In the consumer area, CD and DAT transports often flag 'INVALID' when stopped or paused.

The **'U'** LED indicates activity in the User-bit channel, and is ON unless the channel contains all-zero data.

The incoming Channel Status bits are accumulated into 192-sample blocks, and the most important fields are displayed in the remainder of the DATA panel. More detailed examination is possible through the menu system.

Note that no PARITY bit indication is included in the sub-frame display. This is because an incorrect parity bit is regarded as a carrier violation, and is captured as a biphasic error in the CARRIER panel.

4.2.2. Channel Status display

The Channel Status block is decoded into the various basic indications on the second and third rows of the DATA panel.

The '**PRO**' and '**CONS**' LEDs indicate professional or consumer format Channel Status respectively. Below these LEDs are the '**CRC ERROR**' and '**COPY PRO**' indicators. Note that only a professional status can indicate a CRC error since the consumer status has no CRC, and only a consumer status can indicate Copy Prohibition since no protection mechanism exists in the professional status.

Either '**15/50**'µs emphasis or '**J17**' emphasis can be indicated, and the channel mode can be shown as '**STEREO**', '**2-CHANNEL**', or '**MONO**'. Note that stereo consumer signals are indicated as '**2-CHANNEL**', since this is the actual definition of the mode in the interface standard.

The '**A≠B**' indicator shows that the A and B sub-frames carry differing Channel Status. This is unusual, but is 'legal' in certain circumstances, most often when different alphanumeric labels are included in a professional status, or when channel numbers are used in a consumer status.

5. DSA-1 controls reference

The front panel of the DSA-1 has fourteen control keys which are highlighted in blue, plus the red **[ON]** key. The following section provides detailed descriptions of all the control key functions.

5.1. Menu keys

Below the LCD display is a row of five keys: **[<]**, **[>]**, **[ENTER/NEXT]**, **[EXIT]** and **[EDIT/CLEAR]** which are used to operate the menu system, which is described in section 6.

In general, these keys are used to navigate the menu system as follows: **[<]** and **[>]** keys cycle through the current menu-level options; **[ENTER/NEXT]** is used to select a sub-menu or menu option; **[EXIT]** leaves the sub-menu or menu-option. Within the sub-menu, any or all of the **[<]**, **[>]**, **[ENTER/NEXT]**, **[EXIT]** and **[EDIT/CLEAR]** keys may be used to operate the sub-menu controls.

5.2. Audio monitor controls

The row of four keys to the left of the **[ON]** key are used to control the audio monitoring functions. Monitoring can be by stereo headphones connected to the headphone jack, or using the internal loudspeaker if the **[LS]** key is selected to 'ON'.

The **[LS MODE]** key selects the monitoring mode as follows: In the 'CHAN' setting, the A or B sub-frame **only** is monitored according to the state of the **[CHAN]** key. In the 'BOTH' state, A and B sub-frames are both monitored. The 'LINE' setting allows direct analogue monitoring of the selected input WITHOUT the D-A converter. This is useful for checking for cable-related demodulation effects, or for induced hum, or simply for monitoring analogue signals. The operation of the **[LS MODE]** key is summarised in the table below:

LS MODE	Loudspeaker	HP-L	HP-R
CHAN (A sel):	A	A	OFF
CHAN (B sel):	B	OFF	B
BOTH:	A+B	A	B
LINE:	LINE	LINE	LINE

The monitoring volume is increased and decreased with the [Λ] and [V] keys respectively. These keys feature a delayed auto-repeat function for sweeping the volume up and down. Whenever the volume controls are operated, a numerical readout and bar display of the volume setting appear on the LCD. The operation of the volume control is from 0 (minimum) to 31 (maximum); note that volume setting 25 corresponds to maximum volume before clipping for a fully-modulated sine input. The settings above this represent 12dB 'in hand' for use with ordinary or under-modulated signals.

The monitoring DAC has a 15/50µs de-emphasis capability which can be set to follow the incoming Channel Status indication, or can be over-ridden to ON or OFF. The de-emphasis mode is set in the Settings Menu, as described in section 6.9.1.3.

5.3. I/O selection controls

The row of five keys at the bottom of the panel are used to control the modes of the I/O connections.

5.3.1. DI key

The [DI] key is used to select the connector format for the input under test. Options are 'XLR', 'COAX', or 'OPT' (optical).

5.3.2. TERM key

The [TERM] key connects and disconnects the DI input terminating impedance in the 'LO-Z' and 'HI-Z' states respectively. Normally, the DI input should be terminated except when making measurements 'in-line', via the '⌚' (loop-through) setting of the [DO] key. In this case, the DO connector carries the signal through the DSA-1 to be terminated at its destination. When the [DO] setting is 'RPT' or 'FNC', the termination setting is automatically over-ridden to 'LO-Z', but returns to the setting selected by the [TERM] key when loop-through mode is re-entered.

The termination value is 110Ω for the XLR DI connector, and 75Ω for the BNC DI connector.

5.3.3. CHAN key

The **[CHAN]** key is used to select the desired sub-frame (channel) for various of the unit's functions which operate on a single sub-frame. These include the DATA panel, the Audio Monitoring and Sync O/P sections in '**CHAN**' mode, and some of the menu functions.

5.3.4. DO key

The **[DO]** key selects the function of the XLR and coaxial output (DO) connectors.

The loop-through setting provides a direct un-buffered path from the equivalent input (DI) connector to allow 'in-line' testing of an XLR or coaxial signal subject to its normal termination etc. with minimal insertion effect from the DSA-1.

The '**RPT**' (repeat) setting causes the output at the DO connectors to be a regenerated version of the signal-under-test. The output signal is re-clocked using the DSA-1's ultra-stable PLL, and its amplitude is also regenerated (see section 3.2.). This regeneration feature is useful in determining whether drop-out at an installed receiver is due to jitter and/or attenuation of the received signal: The DSA-1 is placed 'in-line' at the errant receiver, and can be toggled between loop-through and RPT modes to diagnose the failure.

The '**FNC**' (function) setting causes the DSA-1 Signal Generator to be turned on and routed to the DO outputs. As in the '**RPT**' setting, the output timing and amplitude are regenerated and independent of the data input. The Signal Generator parameters are controlled from the menu system, as described in section 6.4.

It will be noticed that when moving from '**RPT**' to '**FNC**' to turn on the Signal Generator, a short delay occurs after the '**FNC**' LED is lit, before the Signal Generator is turned on. Similarly, when moving out of '**FNC**' mode, a delay occurs before the selected loop-through or RPT mode is reinstated. This is necessary because it takes a short while for the DSA-1 to install and remove the Signal Generator mode, and these delays prevent the Signal Generator from being activated when 'passing through' the '**FNC**' setting, for instance when switching repeatedly between loop-through and RPT modes.

Some interaction will be noticed between the **[DO]** '**RPT**' and '**FNC**' modes, the **[SYNC O/P]** '**REF**' mode and certain menu operations (for example the Ref Learn menu function). Variously, these modes may over-ride each other or lock each other out depending on allocation constraints of the DSA-1 hardware.

For example, if the **[DO]** mode is set to **'RPT'** and then **[SYNC O/P]** is set to **'REF'** or the Ref Learn menu is entered, the **[DO]** mode is over-ridden to 'loop-through' since the PLL cannot be used for DO regeneration whilst locked on the Ref Sync signal. **'RPT'** mode is automatically reselected on exiting the over-riding mode, unless the **[DO]** mode has been manually reselected in the mean time.

Additionally, selection of the **'FNC'** mode is prevented during operation of some menu functions (for example the Microscope) and conversely these menus cannot be entered whilst the Signal Generator is switched on.

5.3.5. SYNC O/P key

The **[SYNC O/P]** key controls the source of the `SYNC' test output (see section 3.5.). In **'CHAN'** mode, the SYNC pulse marks the start of the preamble for the A or B sub-frame (the X- or Y-preamble), as selected by the **[CHAN]** key. In **'BLK'** mode, the SYNC pulse marks the Z-preamble at the start of each Channel Status block.

'REF' mode behaves like **'CHAN'** mode, except that the SYNC pulse is referred to the REF SYNC input. In this case, the DSA-1 automatically selects whichever REF SYNC input is in use, and determines whether an IEC958/AES11 or Wordclock reference is connected. In the latter case, the SYNC pulse is referenced to the rising edge of the Wordclock.

The timing of the SYNC output is arranged so that its rising edge slightly precedes the start of the selected preamble. This allows the whole preamble and following data to be easily viewed on an oscilloscope. Note that when the DSA-1 Signal Generator is in use and synchronised to DI, the sync pulse is slightly advanced with respect to the DI data owing to internal timing constraints.

The **[SYNC O/P]** key may interact with some other functions, principally in that **'REF'** mode may be unavailable when certain menu or Signal Generator functions are active; refer also to the note in section 5.3.4.

6. DSA-1 menus reference

The DSA-1 has an LCD display at the top of the control panel and this is used with the [**<**], [**>**], [**ENTER/NEXT**], [**EXIT**] and [**EDIT/CLEAR**] keys to provide an extensive menu system for access to various measurement modes.

In general, these keys are used to navigate the menu system as follows: [**<**] and [**>**] keys cycle through the current menu-level options; [**ENTER/NEXT**] is used to select a sub-menu or menu option; [**EXIT**] leaves the sub-menu or menu-option. Within the sub-menu, any or all of the [**<**], [**>**], [**ENTER/NEXT**], [**EXIT**] and [**EDIT/CLEAR**] keys may be used to operate the sub-menu controls.

As an example, to display the currently-selected Test Sequence, do the following:

First check that you are at the top level of the menu system, if necessary by pressing [**EXIT**] several times. Now press [**>**] to cycle through the top-level menu option until you see the following in the LCD display:

```
Select Mode:  
3. Test Sequence
```

Now press [**ENTER/NEXT**] to enter the 'Test Sequence' menu, and you will see:

```
3. Test Sequence  
3.1. Run
```

Press [**>**] to cycle through the 'Test Sequence' sub-menus, and leave the display set to:

```
3. Test Sequence  
3.2. Select
```

Pressing [**ENTER/NEXT**] now enters the 'Select' sub-menu, and the display shows something like:

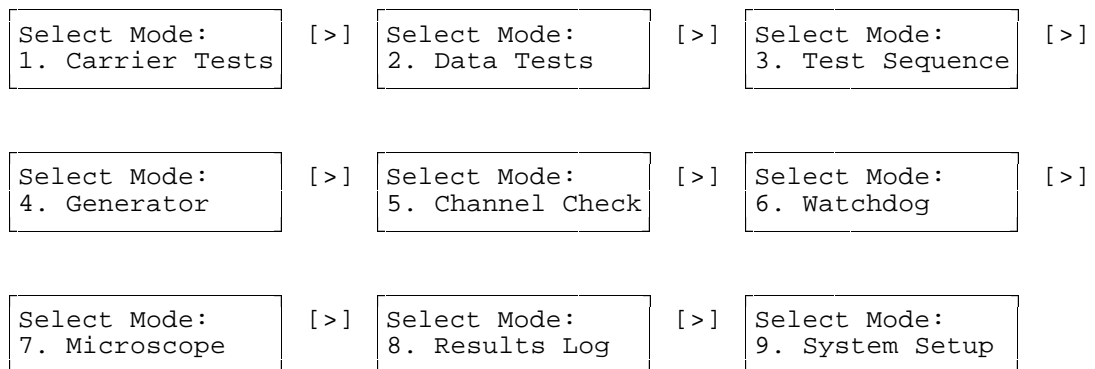
```
Selected: Sys01  
Global Test
```

A different Test Sequence may be shown if the 'Global Test' was not currently selected.

Pressing [**EXIT**] twice returns to the top-level of the menu system.

This example illustrates the general method of navigating the menu system. The following sections provide details of the features available through the menu system.

There are nine options available at the top menu level in version 2.0 of the DSA-1 firmware. These options can be cycled using the [**<**] and [**>**] keys:



When the desired option is displayed, press [**ENTER/NEXT**] to access the selected sub-menu.

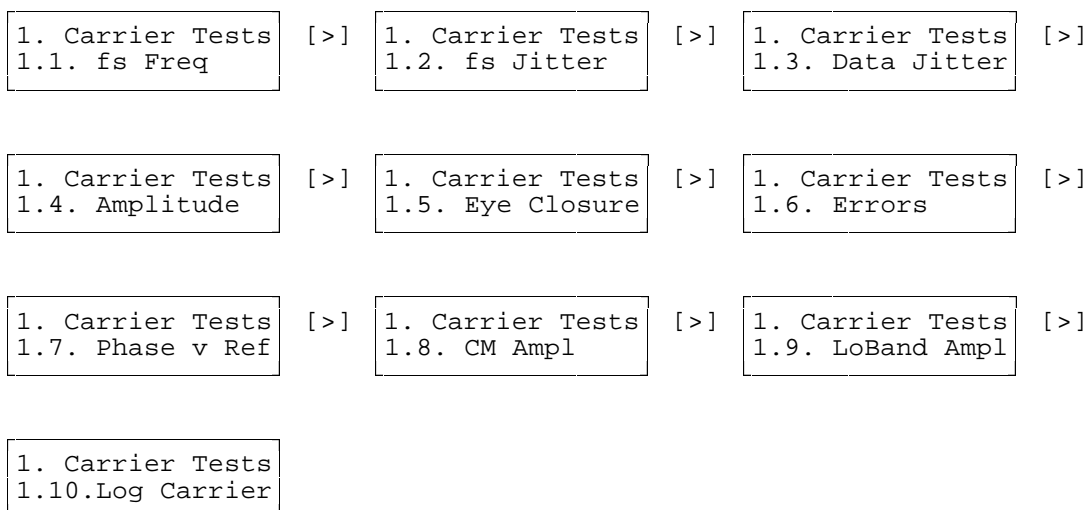
Figure 6.1 overleaf shows a summary of the DSA-1 V2.0 menu options.

1. Carrier Tests		
1.1.	fs Freq	Measures sampling rate
1.2.	fs Jitter	Measures fs (source) Jitter
1.3.	Data Jitter	Measures Data (total) Jitter
1.4.	Amplitude	Measures carrier amplitude
1.5.	Eye Closure	Measures loss of eye width
1.6.	Errors	Displays biphase error count
1.7.	Phase v Ref	Measures ϕ wrt Ref Sync
1.8.	CM Ampl	Measures common-mode ampl
1.9.	LoBand Ampl	Measures sub-20kHz ampl
1.10.	Log Carrier	Runs Carrier Tests to log
2. Data Tests		
2.1.	Audio Meter	Audio bargraph and readout
2.2.	Active Bits	Displays audio-bit activity
2.3.	C-S Fields	Channel Status by field
2.4.	C-S Bytes	Channel Status by byte
2.5.	C-S Idents	Channel Status alpha labels
2.6.	C-S Timecode	Channel Status timecodes
2.7.	Log Data	Runs Data Tests to log
3. Test Sequence		
3.1.	Run	Runs selected sequence
3.2.	Select	Selects desired sequence
3.3.	Log Sequence	Runs selected sequence to log
4. Generator		
4.1.	Sync Source	Selects int/ext sync or DI
4.2.	Edit Modes	GEN1: sets params to edit
4.3.	Audio Func	Selects GEN1/GEN2, & fn
4.4.	Audio Params	ON/OFF, freq, ampl, word
4.5.	Chan Status	GEN1: C-S edits, GEN2: P/C
4.6.	C-S Capture	Loads input C-S to GEN1
4.7.	Valid Bit	GEN1: sets gen Validity bit
4.8.	Jitter Gen	GEN2: gen jitter fn & ampl
4.9.	Log Gen	Writes generator setup to log
5. Channel Check		
5.1.	Run to Log	Logs bit-errors against time
5.2.	Setup	Sets channels, word-length
6. Watchdog		
6.1.	Run to Log	Logs panel LEDs against time
6.2.	Setup	Selects LEDs to be watched
7. Microscope		
Digital gain for monitoring low-level signals, with clip indicators		[NEXT] selects field to adjust [<],[>] adjust selected field
8. Results Log		
8.1.	Start Block	Inserts block marker in log
8.2.	Print Log	Sends log to serial printer
8.3.	Clear Log	Clears contents of log
8.4.	Log Viewer	Views log line by line
8.5.	Blk Manager	Prints, clears individual blks
8.6.	Form Feed	Sends form-feed to printer
9. System Setup		
9.1.	Settings	Adjusts system settings
9.2.	Save Set-up	Saves settings as power-up
9.3.	Defaults	Sets defaults (doesn't save)
9.4.	Ref Learn	Soft-cal for fs-measurement
9.5.	Battery Chg	Starts fast-charge of battery
9.6.	Host Mode	Interacts with host PC

Figure 6.1. - DSA-1 V2.0 Menus Summary

6.1. Carrier tests (menu 1.)

The following sections describe the Carrier tests; these are tests of the electrical and timing parameters of the signal-under-test. There are ten different sub-menus, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

6.1.1. fs Freq (menu 1.1.)

This selection complements the information provided on the Fs sections of the CARRIER display panel, providing more precise details. It measures the sampling-rate of the signal-under-test using the internal TCXR (temperature compensated crystal reference), optionally taking into account calibration data from a previously-connected external reference, obtained in 'Ref Learn' mode (see section 6.9.4.). The frequency and accuracy are displayed together as follows:

fs = 44100.44 Hz
 Acc = +10 ppm

If an external reference has been 'learned' then the '**WRT REF**' indicator on the CARRIER panel is lit, and an asterisk appears to the right of the 'ppm' text the above display. For best accuracy, it is recommended that the DSA-1 be switched on for a few minutes before using 'Ref Learn' and that the unit be similarly warmed up before making measurements.

To return to the 'Carrier Tests' level of the menu system, press [**EXIT**].

6.1.2. fs Jitter (menu 1.2.)

This selection measures interface timing jitter at the sampling-rate, which is principally a parameter of the transmitting equipment. The results of the measurement are displayed as follows:

```
fs Jitter [P-P]
= 8.7ns fc=1k5
```

The reading is taken by continuously measuring the timing offset of a particular transition in the carrier from a reference instant derived by an ultra-stable PLL (phase-locked-loop) within the DSA-1. Processing of the collected data yields a peak-to-peak value for this timing uncertainty, or jitter. The transition chosen for this analysis is as far away as possible from transitions which are subject to data-related variation and so the reading is reasonably independent of program content and cable-related effects, at least until the jitter becomes very marked. This process is inherently self-calibrating, since all temporal measurements are interleaved with calibrations using known time intervals from the internal TCXR. More detail of jitter-related measurement is provided in section 8 and appendix A.

The measurement of jitter is subject to a low-frequency roll-off in the rejection response of the PLL, which means that jitter below the PLL corner frequency is tracked by the PLL, and so cannot be measured. The DSA-1 PLL's corner frequency is displayed in the bottom right-hand corner of this sub-menu, and can be toggled between 700Hz or 1.5kHz using the **[EDIT/CLEAR]** key. The 1.5kHz setting is intended for general use since it has lower intrinsic jitter and so can measure jitter at lower amplitudes; the 700Hz setting should be used specifically to detect low-frequency jitter.

The **[ENTER/NEXT]** key toggles the jitter-measurement units between 'ns' (nanoseconds) and '%UI' (percent of one Unit Interval, see appendix A).

The **[<]** and **[>]** keys may be used to respectively decrease or increase the amplitude of the GEN2 jitter modulator (see section 6.4.8), if it is operating. Thus jitter transfer can be checked without having to continually navigate between menus. Each time the modulator is adjusted, the LCD screen momentarily shows the modulator settings.

Occasional appearance of the '*RANGE' message indicates that the DSA-1 is momentarily having difficulty in tracking the range of rapidly-changing jitter.

To return to the 'Carrier Tests' level of the menu system, press **[EXIT]**.

Note: The 1.5kHz setting of the PLL has been revised from the 3kHz setting provided in the revision B hardware to provide improved low-frequency coverage in the 'standard' setting. This is possible because the intrinsic jitter of the PLL in the revision C hardware is lower than for revision B, and the response flatter.

6.1.2.1. Jitter measurement while the Signal Generator is on

When the Signal Generator is on, Jitter can still be measured if GEN2 mode is selected and if the incoming carrier is phase-locked to the Generator (or vice-versa). If the phase error between input and Generator exceeds 25%, as indicated by the '**BLOCK**' LED, the jitter reading is 'dashed out', just as if the test input were unlocked. This normally makes it quite obvious if the input and Generator are unlocked. If the input and Generator are unlocked but similar in frequency, readings may be unpredictable as the DSA-1 tries to fix onto the 'moving target'.

When the Generator is on, the DSA-1 uses the Generator's Ref Sync as its own timing reference, therefore it is not restricted in low-frequency jitter measurement by the corner-frequency of the PLL. The low-frequency limit in this case is governed by the jitter sampling period and is just under 50Hz. The corner-frequency indicator on the DSA-1 is thus replaced with '<50*'.

6.1.3. Data Jitter (menu 1.3.)

This selection measures interface timing jitter in a similar manner to that described in section 6.1.2. The main difference is that a greater number of transitions are processed, and these are deliberately selected to be those which are modulated by data activity. The results are similarly displayed:

Data Jitttr [P-P] = 8.7ns fc=1k5

This measurement represents the total of source- and cable-related jitter, and so in most situations is principally cable-related; it will increase markedly with increasing cable length and capacitance. A cable which is poor in these respects causes 'intersymbol interference' (or 'pulse smearing'), wherein the timings of transitions in the data are displaced from their ideal instants owing to the effects of preceding transitions or gaps.

If timing for A/D or D/A conversions is derived from the received interface signal, as is frequently the case, it is vital for the conversion equipment to have very good jitter rejection, ideally down to sub-audible frequencies, if unacceptable conversion degradation is not to result. Since very few items of conversion equipment currently have this property, it is important to minimize interface jitter by all means possible. Cabling is the single most important factor in this. Extreme intersymbol interference

transcends mere converter performance, since it may even cause intermittent data recovery at a receiver in a way which may appear program-related.

The Data Jitter reading is itself program dependent, in that it varies according to the patterns of data being transmitted. The Data Jitter for a completely audio-and-Status-black signal should be similar to its fs Jitter. On the other hand, the Eye Closure sub-menu described in section 6.1.5 provides a cable-inclusive jitter measurement largely independent of program or ancillary bit activity. More detail of jitter-related measurement is provided in section 8 and appendix A.

The measurement of jitter is subject to a low-frequency roll-off in the rejection response of the PLL, which means that jitter below the PLL corner frequency cannot be measured. The DSA-1 PLL's corner frequency is displayed in the bottom right-hand corner of this sub-menu, and can be toggled between 700Hz or 1.5kHz using the **[EDIT/CLEAR]** key. The 1.5kHz setting is intended for general use since it has lower intrinsic jitter and so can measure jitter at lower amplitudes; the 700Hz setting should be used specifically to detect low-frequency jitter.

The **[ENTER/NEXT]** key toggles the jitter-measurement units between 'ns' (nanoseconds) and '%UI' (percent of one Unit Interval, see appendix A).

The **[<]** and **[>]** keys may be used to respectively decrease or increase the amplitude of the GEN2 jitter modulator, if it is operating. Thus jitter transfer can be checked without having to continually navigate between menus. Each time the modulator is adjusted, the LCD screen momentarily shows the modulator settings.

Occasional appearance of the '*RANGE' message indicates that the DSA-1 is momentarily having difficulty in tracking the range of rapidly-changing jitter.

Refer to section 6.1.2.1 above for special notes on measuring jitter whilst using the Signal Generator.

To return to the 'Carrier Tests' level of the menu system, press **[EXIT]**.

Note: The 1.5kHz setting of the PLL has been revised from the 3kHz setting provided in the revision B hardware to provide improved low-frequency coverage in the 'standard' setting. This is possible because the intrinsic jitter of the PLL in the revision C hardware is lower than for revision B, and the response flatter.

6.1.4. Amplitude (menu 1.4.)

This selection measures the peak-to-peak differential carrier amplitude for either the co-axial input or the balanced input. The measurement is made regardless of lock state, and is displayed in volts or millivolts according to voltage.

```
Amplitude =  
562mV [P-P]
```

The measurement is made using an absolute peak detection method, and so is subject to inclusion of any ringing or overshoot on the carrier transitions.

When measuring analogue audio signals, note that the low-frequency response only extends down to 300Hz in this sub-menu. For audio frequencies, use the LoBand Amplitude sub-menu which extends down to 28Hz.

To return to the 'Carrier Tests' level of the menu system, press **[EXIT]**.

6.1.5. Eye Closure (menu 1.5.)

The Eye Closure measurement is similar in use to the Data Jitter measurement (see section 6.1.3.) in that it measures intersymbol interference or pulse smearing. However, this selection does so in a special way which is designed to be independent of program or status activity.

This is done by measuring the duration of the first short pulse in the X preamble, and comparing this result with the ideal duration, derived from the measured sampling rate. The reduction in the duration of this single-cell pulse, following a three-cell symbol, is a useful static measure of the intersymbol interference which manifests itself as Data Jitter when program is present. It also usually represents the worst-case eye closure in the data stream since the 3:1 symbol ratio is the worst possible in the AES3 format. This is described further in appendix A.

The result is shown in nanoseconds, also as a percentage of the ideal '1UI' value:

```
Closure @ZeroX =  
10.2ns, 6.3%UI
```

The measurement of eye-width reduction can be made at the zero-crossing point (as in the example shown above) or at a 'slicing level' equivalent to the 200mV P-P point referred to in the interface specification. The selection is made using the **[EDIT/CLEAR]** key.

When measuring Eye Closure whilst the Signal Generator is on, it is necessary that the test input be phase-locked to the Generator (or vice versa) - see section 6.1.2.1.

To return to the `Carrier Tests' level of the menu system, press **[EXIT]**.

Note: The Eye Closure measurement is closely related to the 'Intersymbol' measurement in the version 1.0 firmware. However, the new Eye Closure measurement is more 'worst-case' (and so generally measures higher) since it also takes into account the effect of any fs jitter in narrowing the eye. It is therefore compatible with the eye width as discussed in the interface standard, if the 200mV threshold is used. The zero-crossing threshold is provided for compatibility with version 1.0 firmware.

6.1.6. Errors (menu 1.6.)

This selection displays a count of biphasic violations since the last acquisition of lock. The result can be cleared to zero by pressing the **[EDIT/CLEAR]** key when this sub-menu is selected.

Errors = 132 since last CLEAR

The interface standard dictates that transitions of the carrier always occur at certain times, and never at certain other times. The presence or absence of extra transitions embodies the varying data. A biphasic error occurs when a transition of the incoming carrier occurs where none is allowed, or when a compulsory transition is missed.

A sub-frame parity error is a type of biphasic error since each sub-frame is required to end in the opposite state to which it began, but a biphasic error does not necessarily imply a parity error.

To return to the `Carrier Tests' level of the menu system, press **[EXIT]**.

6.1.7. Phase v Ref (menu 1.7.)

This selection measures the phase of the selected DI input signal with respect to the reference REF SYNC signal. The DSA-1 automatically selects whichever REF SYNC input is in use and also differentiates between Wordclock (SDIF-2) and AES11 format. If a Wordclock-type reference is used, the in-phase state is when the starting transition of the carrier's X-preamble coincides with the rising transition of the Wordclock. The display shows the phase offset in percent (according to the interface standard) or in degrees; the units are selected with the **[EDIT/CLEAR]** key. A positive reading represents the test signal *lagging* the REF SYNC:

Phase wrt Ref = - 1.6%

To return to the 'Carrier Tests' level of the menu system, press **[EXIT]**.

6.1.8. CM Ampl (menu 1.8.)

This selection measures the common mode voltage present for the balanced XLR input format. If either co-axial or optical inputs are selected no result is displayed. The result displayed is the peak-to-peak range of the mid-voltage of pins 2 and 3, with respect to pin 1:

Commonmode Ampl= 70mV [P-P]

This measurement is useful for detection of cabling faults, as further described in section 8.

To return to the 'Carrier Tests' level of the menu system, press **[EXIT]**.

6.1.9. LoBand Ampl (menu 1.9.)

This selection measures the peak-to-peak analogue audio-bandwidth (28Hz to 20kHz) signal voltage on either the balanced or coaxial DI input. If the optical input is selected, no reading is displayed:

Audio-band Ampl= 140mV [P-P]

To return to the 'Carrier Tests' level of the menu system, press **[EXIT]**.

6.1.10. Log Carrier (menu 1.10.)

Entering this sub-menu causes a summary of all Carrier Tests to be sent to the Results Log. The summary is preceded by a description of the currently-selected DI input and its error status (as reported on the top line of the CARRIER panel).

A typical entry might look something like this in the Results Log:

```
Prism Sound DSA-1 Results Log:
-----
```

```
*** BLOCK 01 ***:
```

```
[Carrier Tests:]
```

```

XLR input, 110R,
Locked
fs Jitter [P-P]   =  2.9ns fc=700
Data Jitter [P-P] =  8.5ns fc=700
Amplitude =       3.829 V [P-P]
Closure @ZeroX =  16.6ns, 10.2%UI
Errors =          0      since last CLEAR
Commonmode Ampl=   677mV [P-P]
Audio-band Ampl=   199mV [P-P]
```

The action of performing the whole range of Carrier Tests takes a little while, so the display shows a counting indication of which of the nine tests is currently in progress.

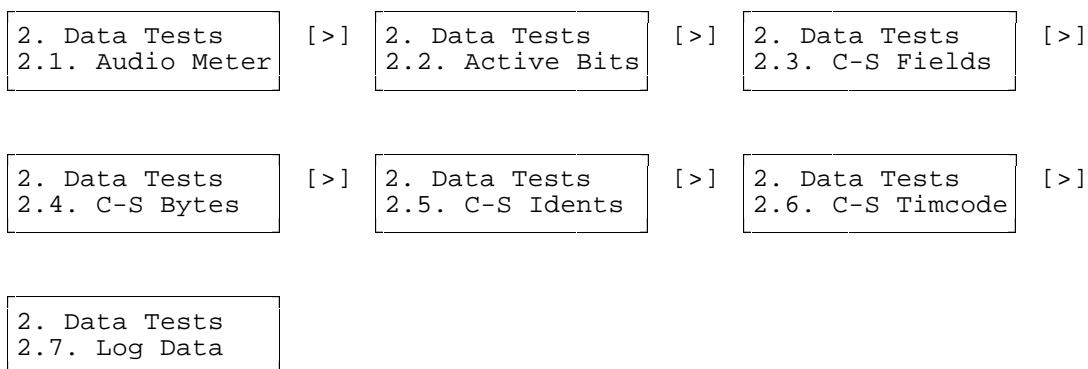
If the Results Log is full, logging is prevented and this is reported by the message:

```
** LOG FULL
```

In this case, the log must be fully or partially cleared as described in section 6.8, which contains a detailed description of the operation of the Results Log.

6.2. Data tests (menu 2.)

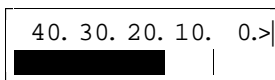
The following sections describe the Data tests; these functions allow examination of the audio and Channel Status data. There are seven different sub-menus, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

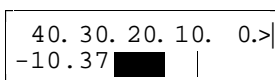
6.2.1. Audio Meter (menu 2.1.)

The DSA-1 audio level meter displays incoming audio peak amplitude in a variety of different ways. In the default mode, a bargraph is shown in the lower half of the display with a dBFS scale in the upper half, as shown below:

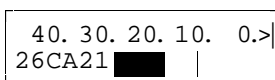


Either the A or the B sub-frame level is displayed according to the selected state of the **[CHAN]** key. A peak-hold bar hovers above the bargraph, holding peak values for a short time before falling back onto the bar. The rightmost segment of the bargraph may show '*' to indicate an overload, which is defined as a single sample containing positive or negative full-scale in the most significant 16-bits. The overload indicator is held for a short time after the overload has occurred.

Pressing the **[ENTER/NEXT]** key turns on a numerical readout of the value of the peak-held sample, shown in dBFS, overlaying the lower part of the bargraph:



Pressing **[ENTER/NEXT]** again changes the format of the numerical readout to a 24-bit hexadecimal notation of the peak-held sample value:



NOTE: If the peak sample value is negative, the hexadecimal readout shows the ONES-COMPLEMENT of the value, i.e. a positive number is displayed which has all the bits of the negative peak value inverted. This is useful since, for active word-lengths less than 24-bits, it allows negative peak samples to be identified as negative, whilst still displaying a positive peak value. For example, a positive peak sample value 16-bit system might be displayed as 5F0300, whereas an equivalent negative peak value would be shown as 5F02FF. **This does not imply activity in the bottom 8-bits.**

Pressing **[ENTER/NEXT]** a third time returns the numerical display to the OFF state.

The **[EDIT/CLEAR]** key can be used to change the operation of the meter to continuous-hold mode. In this mode, the peak value is held by the peak-hold bar indefinitely until it is cancelled by pressing **[EDIT/CLEAR]** again to cancel continuous-hold mode. The overload indicator is also held indefinitely in this mode, and the numerical readout, if turned on, records the value of the held peak.

The **[<]** or **[>]** keys toggle between mono and stereo mode. In stereo mode, both channels are shown at once (the A sub-frame is above the B sub-frame) although the

scale is lost through lack of space. In stereo mode, since one peak detector is shared between both channels, it is possible for transient peak readings to be missed. The numerical-readout, continuous-hold and stereo modes can all be selected independently; stereo mode with numerical readout in dBFS looks like this:

-11.44		
- 9.12		

To return to the 'Data Tests' level of the menu system, press **[EXIT]**.

6.2.2. Active Bits (menu 2.2.)

This selection shows the activity of the 24 bits in the audio and aux fields of the selected sub-frame. In this way, the active word-length of an equipment output can be checked.

Each bit is represented by an arrow symbol, arranged in three byte-wide groups, with the most-significant bit on the left. A double-headed arrow indicates an 'active' bit, i.e. moving or fixed at logic 1; a down-pointing arrow represents an 'inactive' bit fixed at zero. Where bits which become active very briefly the display of the active symbol is 'stretched' so that activity is not missed.

On the top line of the display, there are three hex-format numbers which indicate the bit activity in an alternative form; note that these numbers refer only to bits that are changing and do not represent an individual sample value unless an unchanging audio data-word is being received. A 16-bit active word-length is indicated in this example:

M	FF	FF	00	L
↑↑↑↑↑↑↑↑	↑↑↑↑↑↑↑↑	↓↓↓↓↓↓↓↓		

This test is useful in system-building for avoiding accidental un-dithered truncation when a receiving equipment accepts fewer bits than are output by its sourcing equipment. See section 8.

To return to the 'Data Tests' level of the menu system, press **[EXIT]**.

6.2.3. C-S Fields (menu 2.3.)

This selection allows the incoming Channel Status data to be examined on a field-by-field basis. The status fields are shown in binary form (in place, in their respective bytes) on the upper line of the display, whilst the lower line contains an explanation of the meaning of the data. The Channel Status field interpretation is defined by whether the status is 'professional' or 'consumer', as determined by the state of Channel Status bit 0. The desired field is selected using the [**<**] and [**>**] buttons.

For example, if the emphasis field of a consumer status is selected, the display might show:

CS,B00: ...00...
No emphasis

The byte number (range 0-23) is shown ('B00' above) on the upper line, and those bits of the current byte relevant to the current field are shown as either '1' or '0'. Bits not included in the current field are shown as points. The MS bit is on the left. In the lower line of the display the meaning of the current field is described.

Because of the field interpretation feature, operation of the Channel Status fields menu is quite self-explanatory, however, one or two features are worth noting:

If a consumer status is being examined, the Category Code in the second byte indicates the nature of the source equipment (e.g. CD, DAT etc.) and can have a very large range of values. The DSA-1 recognises a very wide range of Category Codes and interprets them. The MS bit of the Category Code byte is interpreted in conjunction with the copy-prohibit bit in byte 0 and with the Category Code itself to determine SCMS (Serial Copy Management System) status. This interdependence of fields for SCMS is quite involved, and is explained in more depth in section 8 and appendix A.

If a professional status is being examined, an asterisk (*) may appear on the upper line next to the byte indicator. This flags the displayed status field as 'unreliable' as indicated by the unreliability field in byte 22.

If a professional timecode field is being examined, the DSA-1 can decode the data assuming it to be either a binary sample count (as defined in the interface standard) or a BCD (binary coded decimal) value. These modes are indicated by ':' or '.' separators in the timecode display respectively. The mode can be toggled using the [**EDIT/CLEAR**] key.

To return to the 'Data Tests' level of the menu system, press [**EXIT**].

6.2.4. C-S Bytes (menu 2.4.)

This selection allows examination of the incoming Channel Status data on a byte-by-byte basis, without any explanation or decoding. The display format is as follows:

CS,B14:	10111000
CS,B15:	00110110

Two bytes are shown at a time; scrolling through the Channel Status data is performed in 1-byte increments using the [**<**] and [**>**] buttons.

To return to the `Data Tests' level of the menu system, press [**EXIT**].

6.2.5. C-S Idents (menu 2.5.)

This selection displays the Channel Status idents (alphanumeric source and destination labels) as text, if professional format is detected and text idents are present; for example:

Srce ID =	"Bass"
Dest ID =	"MIXR"

Each ident is decoded into a 4-character text string and displayed as above. Null or illegal character codes are displayed as filled blocks.

This sub-menu is available directly from the `Data Tests' menu to avoid having to scroll through the Channel Status each time it is wished to inspect the Channel Status idents.

To return to the `Data Tests' level of the menu system, press [**EXIT**].

6.2.6. C-S Timecode (menu 2.6.)

This selection displays the Channel Status Timecode fields, if professional format is detected and timecodes are present.

LOC= 02:22:33:00
DAY=*11:59:21:00

Each of the timecode fields is decoded into an hours:minutes:seconds format with the units for the last digits dependent on the display mode as follows: the DSA-1 can decode Channel Status timecode data represented as a binary sample count (as defined in the interface standard) or as BCD (binary coded decimal) data. These display modes are indicated by ':' or '.' separators in the timecode display respectively. The display mode can be toggled using the **[EDIT/CLEAR]** key.

If an asterisk (*) is present next to a displayed timecode, that timecode is flagged as 'unreliable' as indicated by the unreliability field in byte 22 of the Channel Status.

This sub-menu is available directly from the 'Data Tests' menu to avoid having to scroll through the Channel Status each time it is wished to inspect the Channel Status timecodes.

To return to the 'Data Tests' level of the menu system, press **[EXIT]**.

6.2.7. Log Data (menu 2.7.)

Entering this sub-menu causes a selection of Data Tests to be sent to the Results Log. The summary is headed by a description of the currently-selected DI input and the current sub-frame selection. This is followed by an indication of audio data bit activity, validity and whether or not the User channel is active. Finally the Channel Status is logged. Only non-zero data is logged in the large 'reserved' area of a consumer Channel Status, whereas all professional fields are logged.

A typical entry might look something like this in the Results Log:

```
*** BLOCK 01 ***:

[Data Tests:]

XLR input, 110R,
Chan A selected:
D23>.....<D0
** INVALID, No user data
Channel Status (A&B identical):
CS,B00: .....1 Profesnal status
CS,B00: .....0. Audio use
CS,B00: ...001.. No emphasis
CS,B00: ..0..... Source locked
CS,B00: 10..... fs = 48.0kHz
CS,B01: ....1000 2-channel mode
CS,B01: 0000.... No user mode ind
CS,B02: .....100 No aux, 24-bit w
CS,B02: 00000... Word len not ind
CS,B03: 00000000 Byte 1 targ vect
CS,B04: 00000000 Not a reference
CS,B05: 00000000 [field reserved]
CS,B06+ 44412D31 Srce ID = "DA-1"
CS,B10+ 56322E30 Dest ID = "V2.0"
CS,B14+*[samples] LOC= 00:00:00:00
CS,B18+*[samples] DAY= 00:00:00:00
CS,B22: ....0000 [field reserved]
CS,B22: 1100.... Unreliable bytes
CS,B23: 11010101 CRC correct
```

If the Results Log is full, logging is prevented and this is reported by the message:

```
** LOG FULL
```

In this case, the log must be fully or partially cleared as described in section 6.8, which contains a detailed description of the operation of the Results Log.

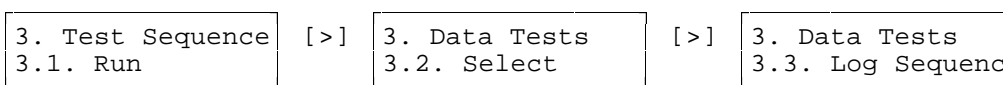
6.3. Test Sequences (menu 3.)

The DSA-1's automated Test Sequences are the key to rapid assessment of the connectivity of digital audio sources. All the individual Carrier and Data tests which can be made by the DSA-1 can be incorporated into automated Test Sequences.

The DSA-1 contains non-volatile memory space for eight Test Sequences. Four of these are permanently programmed and cannot be modified (the 'System' Test Sequences) whereas four can be programmed by the operator (the 'User' Test Sequences).

User Test Sequences are generated by writing a 'program' using the DSA-1 Test Sequence Compiler software package on a PC. The program is then 'compiled' and downloaded into the DSA-1, after which it can be simply executed at the touch of a key. The order of the tests, the limits of acceptability, the displayed messages, and actions on failure can all be chosen by the programmer. Detailed descriptions of how to write and download User Test Sequences are contained in section 7. This section deals only with selection and operation of Test Sequences already resident in the DSA-1.

There are three different sub-menus in the Test Sequences menu, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

6.3.1. Run (menu 3.1.)

Entering the 'Run' sub-menu by pressing **[ENTER/NEXT]** causes the currently-selected Test Sequence to be executed. The progress of the Test Sequence is reported in the LCD display in a manner which corresponds to the programming of the Sequence.

Each Test Sequence is divided into a number of 'Segments'; each Segment contains a series of one or more tests relating to a particular area of investigation of the signal-under-test. The name of each successive segment is displayed in the upper line of the LCD as the segments are run. This results in a 'flicking-through' of the test Segment names as the Test Sequence progresses; long Segments such as jitter or eye-closure tests will cause the display to pause for a few seconds before continuing to the next segment.

If no failures are detected, the LCD displays a message such as:

```
General Pro Test
* Tests complete
```

Alternatively, a failure of any Segment causes further tests to be suspended, and a message is displayed describing the failure. If, for example, System Test Sequence #4 (the Strict Professional Test) is used, and the sampling-rate is incorrectly indicated in the Channel Status, the following failure message is displayed:

```
* FAILED @SEG: 7
fs Conflict
```

In general, the top line of the display indicates which Segment of the test has failed and the bottom line indicates the nature of the fault by displaying a message chosen by the Sequence programmer to explain the problem.

After the Test Sequence has been run to completion or Segment failure, it is possible to re-run the Sequence from the beginning by pressing the **[ENTER/NEXT]** key. Alternatively, the **[<]** and **[>]** keys may be used to select a starting Segment prior to pressing **[ENTER/NEXT]**. The first press of **[<]** or **[>]** after Segment failure moves the starting Segment to the Segment which failed, subsequent presses move backward or forward through the Segments of the Sequence. This allows an offending Segment to be conveniently re-run, or to be passed over. The **[EDIT/CLEAR]** key may be used immediately after completion or Segment failure to reselect the starting Segment to the beginning of the Sequence, prior to subsequent adjustment with the **[<]** and **[>]** keys. This is useful for navigating long Test Sequences.

Whilst in the 'Run' sub-menu, most of the non-menu controls are inactive. These controls are reactivated on leaving the 'Run' sub-menu.

To return to the 'Test Sequence' level of the menu system, press **[EXIT]**.

6.3.2. Select (menu 3.2.)

The 'Select' sub-menu is used to select either a System Test Sequence or a User Test Sequence as the currently-selected Sequence.

The DSA-1 has System Test Sequences, as follows:

Sys01: **Global** (consumer or professional, limited checking of Channel Status)

Sys02: **Consumer** (basic checking, generally equivalent to IEC958 consumer)

Sys03: **General professional** (basic checking, generally equivalent to AES3-1992)

Sys04: **Strict professional** (requires good carrier and full status implementation)

There are also four User Test Sequences available, if these have been loaded.

To select a Test Sequence from this menu, scroll through the list using the [**<**] and [**>**] keys until the desired Sequence is displayed. The Sequence designation is shown in the upper right of the display, with the Sequence title on the bottom line. The titles of User Test Sequences are decided by the Sequence programmer when the Sequence is composed, for example:

Selected: Usr02 Ian's mixer test
--

When the desired Sequence is displayed, leave the 'Select' sub-menu by pressing [**EXIT**]. The selected Test Sequence will remain current until another is selected or until the DSA-1 is powered down.

If any of the User Test Sequences have not been programmed (or have been erased) then the message '[Empty]' appears in place of the title. If the sub-menu is exited with an empty or corrupt User test Sequence selected, Sys01 (the Global Test) is automatically selected instead.

On power-up the DSA-1 selects whichever Test Sequence was current when the system settings were last saved (see section 6.9.2). This allows a preferred Test Sequence to be automatically loaded on power up.

6.3.3. Log Sequence (menu 3.3.)

The 'Log Sequence' sub-menu behaves exactly like the 'Run' sub-menu 3.1. (described above) except that the progression of the Test Sequence is recorded in the DSA-1 Results Log.

A typical Test Sequence session might look something like this in the Results Log:

```
Prism Sound DSA-1 Results Log:
-----

*** BLOCK 01 ***:

[Test Sequence:]

Strict Pro Test

1. Carrier qual    PASSED
2. Pro CS          PASSED
3. fs frequency    PASSED
4. Amplitude       PASSED
5. Jitter          PASSED
6. Eye width       PASSED
7. Channel Stat    **FAILED  CRC error
8. Biphase errs    PASSED
*Tests Complete
```

If the Results Log is full, logging is prevented and this is reported by the message:

```
** LOG FULL
```

In this case, the log must be fully or partially cleared as described in section 6.8, which contains a detailed description of the operation of the Results Log.

To return to the 'Test Sequence' level of the menu system, press **[EXIT]**.

6.4. Generator (menu 4.)

The DSA-1 Generator/Editor is a powerful facility which allows generation of audio test signals, Validity, Channel Status patterns and jitter at the DO outputs. The **[DO]** key must be set to the **'FNC'** position for the Generator/Editor output to be available.

Synchronisation may be internal, or from a balanced or coaxial AES11 source, or from an SDIF-type Wordclock. An 'in-line-edit' mode is available (by setting the Sync Source to 'DI') which allows selective editing of some aspects of an external signal passed through the DSA-1, for example a single Channel Status field could be corrected or some type of jitter superimposed.

The DSA-1 Generator/Editor is of a level of sophistication hitherto unknown in a portable form, indeed many of its features are not widely available even in bench equipment. Such is the versatility of the Generator/Editor that a large number of possible adjustments and modes are provided which the operator should understand in detail if the feature is to be used to its full potential. A block diagram of the Generator/Editor is shown in figure 6.2.

The DSA-1 Generator/Editor is actually based on TWO alternative generator blocks (selected in the Audio Function menu, 4.3.), which are referred to as GEN1 and GEN2. GEN1 is generally the more fully-featured of the two, but 'locks out' certain other features of the DSA-1, principally the Carrier Tests. For this reason, GEN2 is provided as a 'Channel Testing' generator: GEN2 has less versatility than GEN1, but does provide the functionality required for Carrier-related tests, including jitter generation. Most of the DSA-1's Carrier Tests are available with GEN2 operating. GEN1 and GEN2 are contrasted in the table on page 25.

The most important concept to understand from figure 6.2. is how the DSA-1 Generator/Editor can behave as either a Generator or an Editor, and how the audio, Channel Status and Valid bit data can be edited independently of each other if required.

It is easier to first consider the 'in-line editing' situation (when the Sync Source is set to 'DI') using GEN1: The operator can choose to either pass through data from the DI input, or substitute data from the Generator instead. This decision can be made independently for audio, Channel Status and Validity (and in fact independently for each Channel Status field). For example, input audio data and the Valid bit from DI could be set to be passed intact, but some of the Channel Status fields could be 'edited', i.e. provided by the DSA-1 Generator; this might be a typical work-around for a Channel Status compatibility problem. Alternatively, line-up tones could be temporarily superimposed on an interconnect, but with Channel Status and Valid bits passed through as normal.

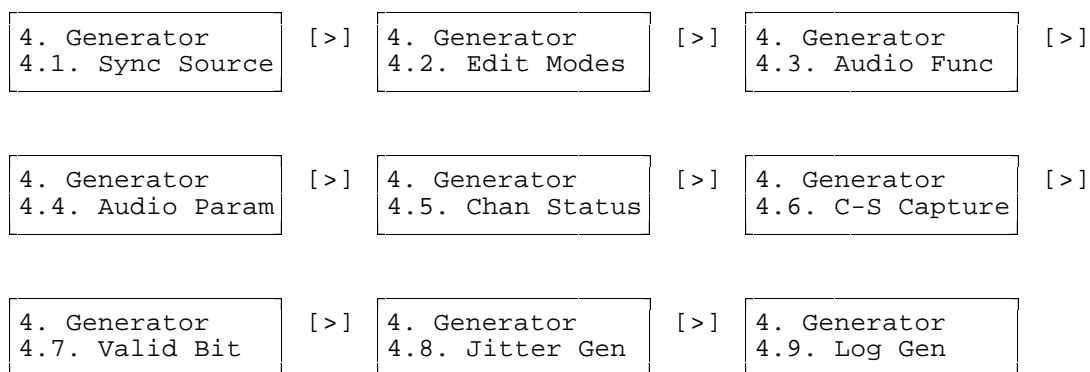
The generation case is really a subset of the 'in-line editing': i.e. when the Sync Source is set to Internal or External Ref Sync rather than DI, the selected Sync Source is

considered to be sourcing audio, Channel Status and Validity 'black' input to the editor (i.e. all zero data), so any fields not set to be edited remain 'black'.

Edited fields may be set to the same values for the A and B sub-frames (channels) or may be set to different values, according to whether the Edit mode is 'TIED' or 'SPLIT'. This applies to audio, Channel Status and Valid bit data.

In the case of GEN2, the possibilities are much more limited: in 'in-line edit' mode, audio, Channel Status and Validity are ALL set to be passed through, and the only parameter to be superimposed is jitter, if required. Otherwise, audio, Channel Status and Valid bit are ALL generated, and the edits are always TIED. In addition, the versatility of the edit data is much more limited for GEN2.

There are nine different sub-menus in the Generator menu, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

HINT FOR SIMPLE USE OF THE GENERATOR: The DSA-1 Generator/Editor is very versatile in its features, but this can lead to problems if you accidentally select an 'advanced' generating or editing mode. If you only need a basic Generator:

- Remember to enable the Generator output by selecting the 'FNC' state of the [DO] key;
- Stick to GEN1 - don't select GEN2 in the Audio Function menu, 4.3.
- Don't use the Edit Modes menu, 4.2.; leave the default settings of TIED, Aud:Y, C-S:Y, V:Y;
- Don't use the 'DI' setting of the Sync Source menu, 4.1.

The table below contrasts the capabilities of DSA-1 Generators GEN1 and GEN2:

Parameter	GEN1 capability	GEN2 capability
Synchronisation Source	Internal (44.1kHz/48kHz), AES11 (XLR or coax), Wordclock , or DI (in-line edit mode)	Internal (44.1kHz/48kHz), AES11 (XLR or coax), Wordclock , or DI (in-line edit mode)
Edit Modes	Audio, Channel Status, Valid bit independently generated or transparent, channels Tied or Split	Audio, Channel Status, Valid bit always generated and channels Tied, except transparent in 'in-line' mode
Audio Functions	Sine, Square, Triangular, Ramp, Pulse (with variable duration and polarity), Jtest	fs/4 sine, PRS (noise), Jtest
Audio Parameters	On/off, Word-length (12-24), Frequency, Peak amplitude	On/off
Channel Status	All fields may be generated or edited independently	Professional or Consumer
C-S Capture facility	Incoming Channel Status may be captured as edit data	N/A
Validity Bit	Editable	Fixed at 0, Valid
Jitter Generation	N/A	Function: PRS-L, PRS-H, fs/8, Cable simulation Amplitude: 1/8 UI steps
Carrier Tests possible	None	All except fs Freq, Phase v Ref
Data Tests possible	All	All except Audio Meter

WARNING: The DSA-1 generator is capable of producing full-scale digital test signals. Care should be exercised when testing systems with monitoring headphones or loudspeakers in order to avoid personal injury or damage to equipment.

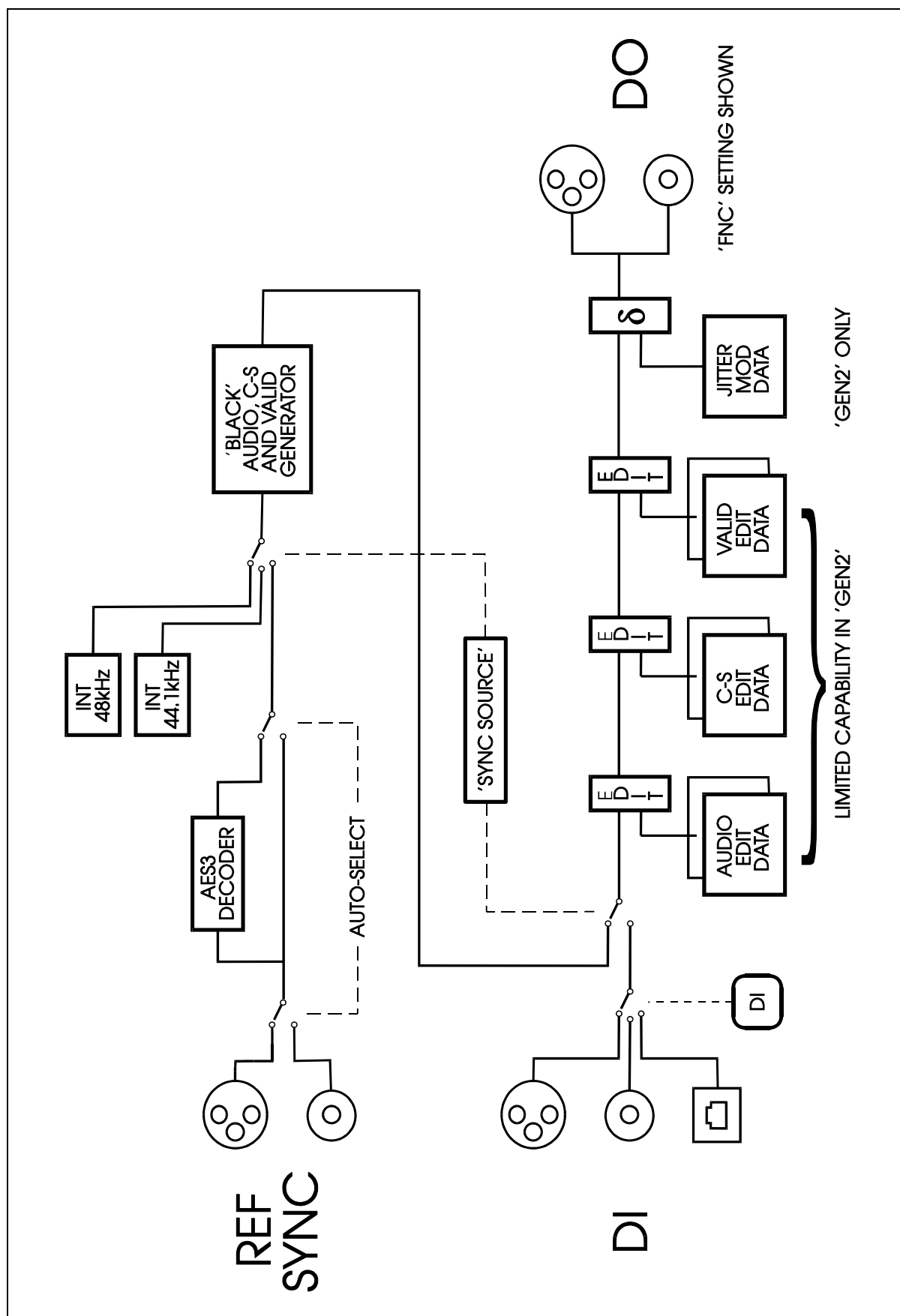


Figure 6.2. - Generator/Editor simplified block diagram

6.4.1. Sync Source (menu 4.1.)

This sub-menu allows the synchronization source of the Generator to be selected, as shown in figure 6.2. and described at the beginning of section 6.4.

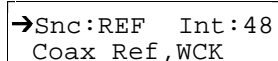
The display format is:

A screenshot of the DSA-1 display showing the Sync Source menu. The text '→Snc:INT Int:48' is displayed within a rectangular border. The arrow points to 'Snc', indicating it is the selected parameter for adjustment.

The flashing arrow symbol indicates which parameter is selected for adjustment, and this selection is cycled using the **[ENTER/NEXT]** key. The **[<]** and **[>]** keys are used to adjust the selected parameter by stepping through the available settings.

In 'INT' (internal) mode, the DSA-1 generator is 'clock master', i.e. it is not synchronized to any external reference. The sample rate of the internal reference is selected in the second field as either 48kHz or 44.1kHz. The precision of the internal reference is derived from the same TCXR as is used to measure sample frequencies in Menu 1. This is nominally $\pm 3\text{ppm}$ as standard, but can be $\pm 0.5\text{ppm}$ to special order.

In 'REF' (Ref Sync) mode, the XLR or coaxial Ref Sync input is used. In this case, the DSA-1 automatically senses the appropriate Ref Sync input, and determines whether an AES11 reference or Wordclock is applied. The result of this sensing is shown on the bottom line of the display when Ref Sync is selected, for example:

A screenshot of the DSA-1 display showing the Sync Source menu in REF mode. The text '→Snc:REF Int:48' is displayed on the first line, and 'Coax Ref,WCK' is displayed on the second line, all within a rectangular border. The arrow points to 'Snc'.

In 'DI' mode, the currently-selected DI signal is used as the generator clock reference. This mode is intended for 'in-line' editing applications, wherein the audio, Valid bit or individual Channel Status fields can be passed through from the DI input if desired, or can be over-ridden by the DSA-1 Generator.

If DI mode or Ref Sync mode is selected but no appropriate input signal is detected, the generator defaults to internal sync at the selected rate. The selected sync mode is adopted as soon as an appropriate reference is detected.

The **[EDIT/CLEAR]** key can be used to select the default value of the currently-selected parameter, which are Internal sync and 48kHz.

To return to the 'Generator' level of the menu system, press **[EXIT]**.

6.4.2. Edit Modes (menu 4.2.)

The functionality of the Edit Modes sub-menu is mainly restricted to GEN1 mode, as selected in the Audio Function sub-menu, 4.3.

The display format is:

```
Editing [TIED]  
→Aud:Y C-S:Y V:Y
```

The flashing arrow symbol indicates which parameter is selected for adjustment, and this selection is cycled using the **[ENTER/NEXT]** key. The **[<]** and **[>]** keys are used to adjust the selected parameter by stepping through the available settings.

The TIED/SPLIT field determines whether all audio, Channel Status and Validity settings which will be made in the other sub-menus will apply to both sub-frames (channels), or just to the current sub-frame as set by the state of the **[CHAN]** key. The SPLIT state allows the A and B channels to be set differently in any or all of these parameters as desired. Note that switching between these two settings causes all sub-frame A parameters to be copied to sub-frame B.

The remaining fields are used to globally enable editing for audio, Channel Status and Validity data respectively. In DI Sync mode, switching any of these fields to 'N' disables generation of the respective data (as controlled in the other sub-menus) and reverts to passing through the DI data. In Internal or Ref sync modes, this simply has the effect of globally 'blacking' the respective data.

These concepts are described in more detail at the start of section 6.4 and in figure 6.2.

The **[EDIT/CLEAR]** key can be used to select the default value of the currently-selected parameter, which are TIED and all fields transparent.

When GEN2 is selected, edits are always TIED. When in 'in-line editing' mode, all field edit selections are forced to 'N', otherwise all are forced to 'Y', as described at the beginning of section 6.4. Since no adjustments are possible in GEN2 mode, the flashing arrow is disabled.

To return to the 'Generator' level of the menu system, press **[EXIT]**.

6.4.3. Audio Function (menu 4.3.)

The Audio Function sub-menu is used to select which of GEN1 or GEN2 is active, and what function (or waveform) is selected for that mode.

The display format is:

```
→A&B:Sine  Gen1
```

The flashing arrow symbol indicates which parameter is selected for adjustment, and this selection is cycled using the **[ENTER/NEXT]** key. The **[<]** and **[>]** keys are used to adjust the selected parameter by stepping through the available settings.

The 'A&B' legend shows that both sub-frames (channels) are subject to adjustment, since in GEN1 mode each sub-frame could contain a different audio generator function if the Edit Mode were set to SPLIT. In that case, the 'A&B' indicator would be replaced by 'ChA' or 'ChB' according to the setting of the **[CHAN]** key.

Available functions are:

For GEN1: Sine, Square, Triangle, Ramp, Pulse or Jtest;

For GEN2: fs/4 sine, PRS, Jtest;

The GEN1 Triangle function is a symmetrical triangular wave; the Ramp function is a falling ramp. The Pulse function is a narrow pulse of programmable width and polarity, repeated every 192 samples. If the Pulse function is selected, the flashing arrow can be moved onto the bottom line of the display as shown below, allowing the duration to be set as 1, 2, 3 or 4 samples, and the polarity to +going, -going, or ±going.

```
A&B:Pulse  Gen1
→Pol:+      Smps:2
```

The Pulse waveform is especially useful for testing meter-responses, particularly overload detectors which are often activated by a number of successive full-scale samples. The Pulse function at 0dBFS amplitude peaks at plus (and minus) full-scale.

The Jtest function is a proprietary test pattern specially formulated to induce worst-case Data Jitter over a long or poor quality cable. The operation of the Jtest is described in section 8.

The GEN2 fs/4 sine function is a fixed frequency sine-wave at fs/4, i.e. 12kHz at fs=48kHz. GEN2 is essentially the 'Channel Testing' generator, so a high-frequency tone is included to allow missed or repeated samples, or remedial interpolation, to be easily heard.

The GEN2 PRS (pseudo-random-sequence) function is a white-noise-like signal, which can also be used as the identifiable test-pattern in Channel Check mode, as described in section 6.4.9, 6.5.

To return to the 'Generator' level of the menu system, press **[EXIT]**.

6.4.4. Audio Parameters (menu 4.4.)

The Audio Parameters sub-menu controls the amplitude, frequency, word-length and ON/OFF state of the Generator/Editor.

The display format is:

→A&B:ON	-14dB
Wrd:24b	1000Hz

The flashing arrow symbol indicates which parameter is selected for adjustment, and this selection is cycled using the **[ENTER/NEXT]** key. The **[<]** and **[>]** keys are used to adjust the selected parameter by stepping through the available settings.

The 'A&B' legend shows that both sub-frames (channels) are subject to adjustment, since in GEN1 mode each sub-frame could contain different audio parameters if the Edit Mode were set to SPLIT. In that case, the 'A&B' indicator would be replaced by 'ChA' or 'ChB' according to the setting of the **[CHAN]** key.

The word-length can generally be adjusted from 12 to 24 bits. The DSA-1 Generator is undithered, but is rounded and truncated to the selected word-length.

Where a function's peak-sample amplitude is adjustable, this is possible between 0dBFS and -120dBFS. 1dBFS steps are provided down to -16dBFS in order to cater for a range of standard line-up levels. Note that amplitude cannot be adjusted for the GEN1 Jtest function, nor for GEN2 functions.

Where a function's frequency is adjustable, this is possible between $f_s/192$ and $f_s/2$ (e.g. 250Hz to 24kHz at $f_s = 48\text{kHz}$) in arbitrary steps. There is also a DC setting where the positive peak amplitude sample is output continuously. Note that frequency is not adjustable for GEN1 Pulse or Jtest functions, nor for GEN2 functions.

The **[EDIT/CLEAR]** key can be used to set the default value of the currently-selected parameter, which are OFF, 16-bit, -14dBFS and 1kHz. -14dBFS is chosen because it is a common line-up level, being +4dBu in systems where +18dBu=0dBFS.

Depending on which generator and function are selected, some of the Audio Parameters settings may be fixed or unavailable. In this case, the flashing arrow cannot be moved to that setting, and the setting may be blanked out. In particular,

GEN2 functions have fixed 24-bit word-length and fixed amplitude, with only the ON/OFF setting adjustable.

Remember that the Audio output of the Generator is only presented at the Generator output so long as Audio editing is set in the Edit Modes sub-menu. If not, the Audio output is 'black' or transparent depending on the selected Sync Source.

6.4.5. Channel Status (menu 4.5.)

The Channel Status sub-menu allows adjustment of the Generator/Editor's Channel Status output. For GEN1, each Channel Status field is edited and displayed separately whereas for GEN2, only a basic Professional or Consumer Channel Status can be selected.

In the case of GEN1, the display is broadly similar to the 'C-S Fields' display (in sub-menu 2.3.), but it has a number of different modes since each field can be independently set to EDIT mode, or can be left TRANSPARENT.

In general, the LCD shows the state of the currently-selected Channel Status field, with the binary value of the field shown 'in-place' within its byte on the top line, and disassembled to explain its meaning on the bottom line. The current field selection is cycled using the [**<**] and [**>**] keys:

RxA 00: 10.....
fs = 48.0kHz

Each Channel Status field is individually set to be edited or to remain transparent. If the currently-selected field is TRANSPARENT, the top-left of the screen shows either 'RxA', 'RxB' or 'blk', and the display reflects respectively the A or B channel of the received DI data, or the 'black' (all zeroes) of the reference, according to the selected Sync Source. If the currently-selected field is EDIT, the top-left indicator is replaced with a flashing 'ChA', 'ChB' or 'A&B' indicator showing which output channels are currently being edited, as determined by the TIED/SPLIT Edit Mode and the state of the [**CHAN**] key; the display reflects the edit value.

Fields are toggled between TRANSPARENT and EDIT using the **[EDIT/CLEAR]** key. The field shown below is in EDIT mode (note that the 'A&B' indicator is flashing):

```
A&B 00: 01.....  
fs = 44.1kHz
```

A field which is set to EDIT may also be in MODIFY mode, wherein the **[<]** and **[>]** keys are used to adjust the edit value rather than to cycle to the previous or next field as normal. MODIFY mode is set and cleared the **[ENTER/NEXT]** key, and is signified by the usual flashing arrow next to the edit value on the upper line of the display. If an attempt is made to enter MODIFY mode when the current field is TRANSPARENT, the field is automatically put into EDIT first. The field shown below is in MODIFY mode (note that both the 'A&B' and arrow indicators are flashing):

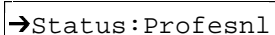
```
A&B 00:→01.....  
fs = 44.1kHz
```

A few fields operate singularly in Professional Channel Status mode (i.e. when byte 0, bit 0 is incoming as, or edited to, a '1'):

- The **alphanumeric source and destination label** field edits are adjusted one character at a time; in this case, the **[ENTER/NEXT]** key cycles from NON-MODIFY, to MODIFY-CHAR1, to CHAR2, to CHAR3, to CHAR4 to NON-MODIFY. Only characters of value 0x00 or >= 0x20 can be selected.
- The **timecode** field edits are adjusted in a similar way to the alphanumeric label fields described above, but cycling from sub-seconds, to seconds, to minutes, to hours. The DSA-1 cannot inject dynamic timecodes, but only static patterns in either sample-count or BCD mode (as set in sub-menu 2.3. or 2.6.). Even so, a static timecode may still be useful in testing timecode receiving devices. Note that wrapping adjustments on one sub-field carry to the sub-field above in sample-count mode but not in BCD mode. In either mode, edit values beyond midnight are not allowed.
- The **CRC** field always injects correct CRC whenever it is in EDIT mode; i.e. the CRC value cannot be edited directly.

Note that in SPLIT mode, different Channel Status patterns can be generated for the two output channels.

If GEN2 is selected, the Channel Status can only be set to a simple Consumer or Professional pattern. The display format is:



```
→Status:Profesnl
```

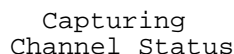
The Professional and Consumer patterns are:

Professional:	Audio use; No emphasis; Source locked; fs as CARRIER panel indication; Stereo; No user data; 24-bit samples; Not a reference; No source or destination IDs; No timecodes; All bytes reliable; CRC correct;
Consumer:	Audio use; No emphasis; Copy permitted; 2-channel; Mode-0; Category code: General; No source or destination numbers; fs as CARRIER panel indication, grade II;

To return to the 'Generator' level of the menu system, press **[EXIT]**.

6.4.6. C-S Capture (menu 4.6.)

On entering the C-S Capture sub-menu, the DSA-1 checks that the currently selected DI is locked (i.e. the CARRIER panel 'FAIL' LED is not lit). If so, the following message is displayed:



```
Capturing  
Channel Status
```

This announces that the Channel Status pattern currently on the selected DI input has been copied into the Channel Status edit buffer for GEN1, and that each Channel Status field has been set to 'Edit' as described in the previous section). This feature

allows the Generator Channel Status to be 'cloned' from a convenient or working source equipment, or used to over-ride the output of another equipment using 'in-line editing' mode.

Note that if the captured Channel Status is in the professional format, a correct CRC is generated even if the incoming CRC was incorrect. Also, the timecode fields of a captured professional Channel Status are set transparent.

If the selected DI input is unlocked (Carrier failed), then the GEN1 edit buffer is cleared instead. This provides a quick method of clearing all fields of the edit buffer. The message in this case is:

Clearing
Channel Status

NOTE: The capture or clearing action described above is always executed, even if GEN2 is currently selected instead of GEN1.

After capturing or clearing is completed, the menu system automatically returns to the 'Generator' level.

6.4.7. Valid Bit (menu 4.7.)

The Valid Bit sub-menu displays the current state of the Generator Valid Bit. The display format is:

→A&B:Valid bit=0
(valid)

If GEN1 is selected, the flashing arrow appears in the upper left-hand corner to show that the [**<**] and [**>**] keys can be used to adjust the state of the Valid bit. If GEN2 is selected, the Valid bit is always set to 0 (valid) and so no adjustment is possible.

The 'A&B' legend shows that both sub-frames (channels) are subject to adjustment, since in GEN1 mode each sub-frame could contain different Valid bits if the Edit Mode were set to SPLIT. In that case, the 'A&B' indicator would be replaced by 'ChA' or 'ChB' according to the setting of the [**CHAN**] key.

Remember that the settings of this sub-menu are only presented at the Generator output so long as Validity editing is selected in the Edit Modes sub-menu. If not, the Validity output is always 'black' or transparent depending on the Sync Source.

To return to the 'Generator' level of the menu system, press [**EXIT**].

6.4.8. Jitter Generator (menu 4.8.)

This sub-menu allows interface jitter to be added to the Generator/Editor output. The amplitude and the function (waveform) of the applied jitter can be adjusted:

```
→Jitter Fn:PRS-H
P-P:UI/8 ,20ns
```

The flashing arrow symbol indicates which parameter is selected for adjustment, and this selection is cycled using the **[ENTER/NEXT]** key. The **[<]** and **[>]** keys adjust the selected parameter by stepping through the available settings.

GEN2 must be selected in the Audio Function sub-menu for jitter generation to be available. Jitter cannot be applied to GEN1; if GEN1 is selected, the display shows that no jitter is applied, and the flashing arrow does not appear.

The peak-to-peak jitter amplitude is adjusted on the bottom line from zero to $3UI/8$ (about 61ns at $f_s = 48\text{kHz}$) in steps of $UI/8$ (about 20ns at $f_s = 48\text{kHz}$). The actual figure in nanoseconds is also displayed if the current sampling frequency is one of the standard rates. If the current sampling frequency is 32kHz (or nearby), the peak-to-peak jitter step size becomes $UI/12$ (about 20ns at $f_s=32\text{kHz}$).

The jitter function is selected on the top line from the following options:

- | | |
|--------|---|
| PRS-L: | Pseudo-random (white noise) jitter, with frequencies confined to approximately audio bandwidth. This is useful for jitter tolerance testing as well as analysing jitter rejection in converter devices. |
| PRS-H: | Pseudo-random (white noise) jitter, with frequencies up to $128 \cdot f_s$. This is useful for jitter tolerance testing. |
| fs/8: | Sinusoidal jitter at $f_s/8$. This is useful for jitter tolerance testing as well as analysing jitter rejection in converter devices. |
| Cable: | Simulation of Data Jitter induced by a long or poor quality cable. Short-duration pulses on the interface tend to be shortened and long-duration pulses lengthened. |

The use of these jitter functions is described in more detail in section 8.

If the Sync Source is set to DI, the selected jitter is applied 'in-line' to the signal applied to the currently-selected DI input.

Note that the possible amplitudes and functions of the applied jitter are quite limited in the DSA-1. The Prism Sound JM-1 is a high-quality analogue jitter modulator which

allows an analogue input to modulate the applied jitter with continuously-variable amplitude up to 500ns. A range of built-in functions are also available.

To return to the 'Generator' level of the menu system, press **[EXIT]**.

6.4.9. Log Generator (menu 4.9.)

On entry, the LCD shows:

```
Logging...
```

The current settings of the Generator/Editor are written to the Results Log. A typical result might be:

```
Prism Sound DSA-1 Results Log:
```

```
-----
```

```
*** BLOCK 01 ***:
```

```
[Gen/Ed Setup:]
```

```
Snc:INT Int:48
Editing [TIED] : Aud:Y C-S:Y V:Y
```

```
A&B:Sine Gen1
A&B:ON -14dB Wrd:24b 1000Hz
```

```
A&B 00: .....1 Profesnal status
A&B 00: ...001.. No emphasis
A&B 00: 10..... fs = 48.0kHz
A&B 01: ....0010 Stereo mode
A&B 02: 00001... 16-bit word len
A&B 06+ 44534131 Srce ID = "DSA1"
A&B 23: [inject] CRC correct
```

```
A&B:Valid bit=0 (valid)
Jitter Fn:----- P-P:OFF
```

If the Results Log is full, logging is prevented and this is reported by the message:

```
** LOG FULL
```

In this case, the log must be fully or partially cleared as described in section 6.8, which contains a detailed description of the operation of the Results Log.

After logging is completed, the menu system automatically returns to the 'Generator' level.

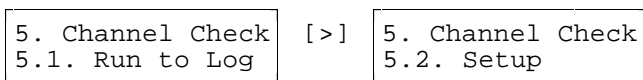
6.5. Channel Check (menu 5.)

The DSA-1 Channel Check mode allows continuous bit-for-bit checking of a digital audio channel, i.e. every bit of every sample is verified to be correct. The channel may be as simple as a cable, or as complex as a satellite link. The delay through the channel is not relevant, and as such it is possible for the channel to be a storage device such as a digital tape or disk recorder. If the ends of the channel are geographically remote, the test pattern can be generated by one DSA-1 and checked by another. The test may be run unattended, for as long a period as required, with any errors being automatically logged both on the display and in the Results Log.

The Channel Checker can test either sub-frame (A or B channel) or both, and the check can be made on 16-bit, 20-bit or 24-bit data. The test signal is the pseudo-random-sequence (PRS) waveform of the DSA-1's GEN2 (see section 6.4), which is similar to white noise at 0dBFS (full-scale) peak amplitude. The test signal contains different data for the two sub-frames.

WARNING: The DSA-1 generator is capable of producing full-scale digital test signals. Care should be exercised when testing systems with monitoring headphones or loudspeakers in order to avoid personal injury or damage to equipment.

There are two different sub-menus in the Channel Check menu, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

6.5.1. Run to Log (menu 5.1.)

Before running the Channel Checker to Log, it is advisable to use sub-menu 5.2. to verify initial reception of an error-free data pattern as described in the next section. When this has been verified, logging can begin by entering sub-menu 5.1.

On entering this mode, the display shows the following:

```
000# 000:00:10.4  
Locked on seq
```

The counting timecode in the upper right shows the time since the logging was started, and the number in the upper left shows how many errors have been detected so far, in this case none.

The message on the bottom line indicates that the PRS data pattern has been successfully identified at the DSA-1's input. If all is not well, you may have some other message, most likely reporting that the Generator mode is incorrectly set or that the input is unlocked. If you have a problem like this, use sub-menu 5.2. to solve it before you recommence logging.

Once the test pattern is correctly identified, any subsequent error or failure results in an 'Event' being stored in the Results Log and the Event count in the top left-hand corner being incremented. A description of the Event appears on the bottom line of the display. If it is a long-term Event, such as loss of carrier lock or loss of the PRS data, the message remains for as long as the condition persists. If the Event is transient, such as a sample data error, it is reported momentarily after which the 'Locked on seq' message returns.

The **[EDIT/CLEAR]** key clears the Event count to zero, restarts the timer, and deletes the Results Log back to the Channel Check Item Marker. This effectively recommences the Channel Check logging operation. Left to itself, the timer wraps to zero after 1000 hours.

The Channel Check display mode described above is called 'Summary' mode, since it shows how many Events have been logged, how long the test has been running, and the current fault status. An alternative display mode is available in Channel Check mode, which can be entered by pressing the **[ENTER/NEXT]** key. In this display mode (the 'Event Viewer' mode), a list of all Events so far logged can be scrolled through using the **[<]** and **[>]** keys. An Event is shown like this in Event Viewer mode:

```
005. 000:29:09.2  
* CARRIER FAIL
```

Alternatively, If no Events have yet been logged, the following message is displayed:

`nothing logged`

Entering Event Viewer mode does not interrupt the progress of the test, i.e. further Events are logged irrespective of whether the display is in Event Viewer or Summary mode.

The **[EXIT]** key terminates the logging operation and leaves Channel Check Run to Log mode. Note that an '<END OF TEST>' event is placed in the log at this point in order that the duration of the test is recorded.

After running the Channel Checker, the format of the Results Log is:

Prism Sound DSA-1 Results Log:

*** BLOCK 01 ***:

[Channel Check:]

001.	000:05:04.1	Sample Error A
002.	001:20:10.6	Sequence Fail
003.	001:20:10.9	Sequenc ReLock
004.	001:32:12.7	Sample Error B
005.	002:45:13.8	<END OF TEST>

It is possible to set up the DSA-1 so that Channel Check Events are accompanied by the transmission of an Event-specific serial code from the RS-232 port. This is done by enabling 'Alarms' in the System Settings sub-menu as described in section 6.9.1.4. If Alarms and Key Beeps are both enabled, each Event is also accompanied by a beep from the DSA-1's loudspeaker.

Refer to appendix A for a list of possible Events in Channel Check mode, and their Alarm codes.

6.5.2. Setup (menu 5.2.)

The Channel Check Setup mode is used to configure the Channel Checker and to verify that everything is set up properly before running the Channel Checker to Log. The display format is:

```
Specify channel:
→ChA  24b  -OK-
```

The bottom line of the display has two adjustable fields, the sub-frame (channel) selection and the audio word-length, as well as an indication at the right of the current status of the Channel Check receiver.

The flashing arrow highlights the currently-selected field, which is adjusted using the [**<**] and [**>**] keys. The [**ENTER/NEXT**] key is used to select the desired field.

Channel Check mode can work on either the A or B sub-frame, or both (A&B). Obviously the single-channel mode should be used if the data path to be tested has only a single-channel capability. When the single-channel mode is selected, the [**CHAN**] key at the bottom of the DSA-1 keypad determines which channel is to be tested.

The audio word-length to be checked can be set to 16, 20 or 24 bits. This should be set to match the capacity of the data path to be tested or, if the data path has some intermediate capacity, to the next smaller value. Setting this value larger than the capability of the data path to be tested results in continuous failure of the Channel Check receiver to match the test pattern.

The '-OK-' indication shown in the example above shows that the Channel Check receiver is successfully receiving the test pattern over the channel(s) and word-length selected. At this point, the Run to Log mode described in the section above can be successfully entered.

If some other message is shown in the lower right-hand corner of the display, then the test setup is incorrect in some way.

'GEN!' indicates that the generator mode is incorrect for Channel Check operation - GEN2 must be selected, set to the 'PRS' function, turned ON, and the [**DO**] key setting must be '**FNC**'. Note that the Generator must be set as above even if the DSA-1 is only being used to check the test pattern and not to generate it (e.g. during remote or off-tape testing);

'----' indicates that no valid AES3 carrier is detected at the selected test input, i.e. carrier FAIL. Either the sourcing equipment is operating incorrectly or the wrong test input is selected on the [**DI**] key;

'ERRS' indicates that the incoming data pattern does not match the PRS over the selected word-length. This might be because the word-length is set to a longer value than the capability of the data path under test, or it may be that the data is being muted or modified somewhere. Note that any signal processing at all in the data path under test (e.g. gain adjustment, re-dithering, EQ etc.) will cause the test pattern to be corrupted. The DSA-1 monitoring loudspeaker can be used to quickly verify that the PRS is present at the test input;

To return to the 'Channel Check' level of the menu system, press **[EXIT]**.

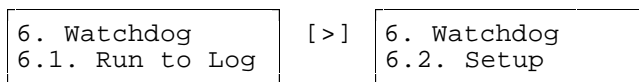
6.6. Watchdog (menu 6.)

Watchdog mode monitors selected LEDs on the DSA-1 CARRIER and DATA panels and details any activity, with time-stamps, in the Results Log. This mode is particularly useful in tracking down intermittent system faults; for example occasional biphasic errors, brief fluctuations in sample-clock accuracy or changes in Channel Status may cause serious operational problems but are usually difficult to isolate.

Setup mode is used to define the particular LEDs which are of interest, after which the Run to Log mode can be left unattended for as long as necessary to log any changes. By using the DSA-1 in 'loop-through' mode, in-line operation of the Watchdog mode is possible during normal operation of an installation.

Watchdog mode can also be configured to flash any LEDs which have been active, however briefly, since the start of the test whether they were selected for logging or not.

There are two different sub-menus in the Watchdog menu, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

6.6.1. Run to Log (menu 6.1.)

Before running the Watchdog to Log, it is necessary to use sub-menu 6.2. to 'alarm' the desired LEDs on the CARRIER and DATA panels as described in the following section (unless this information has previously been entered and saved with the system settings). When this has been done, logging can begin by entering sub-menu 6.1.

On entering Run to Log mode, the display shows the following:

```
000# 000:00:10.4
Watchdog running
```

The counting timecode in the upper right shows the time since the logging was started, and the number in the upper left shows how many errors have been detected so far, in this case none.

Any change in the state of an 'alarmed' LED results in an 'Event' being stored in the Results Log and the Event count in the top left-hand corner being incremented.

The **[EDIT/CLEAR]** key clears the Event count to zero, restarts the timer, and deletes the Results Log back to the Watchdog Item Marker. This effectively recommences the Watchdog logging operation. Left to itself, the timer wraps to zero after 1000 hours.

The Watchdog display mode described above is called 'Summary' mode, since it shows how many Events have been logged, how long the test has been running. An alternative display mode is available in Watchdog mode, which can be entered by pressing the **[ENTER/NEXT]** key. In this display mode (the 'Event Viewer' mode), a list of all Events so far logged can be scrolled through using the **[<]** and **[>]** keys. An Event is shown like this in Event Viewer mode:

```
019. 001:49:29.5
Biphase Error
```

Alternatively, If no Events have yet been logged, the following message is displayed:

```
nothing logged
```

In Event Viewer mode, the normal operation of the CARRIER and DATA panel LEDs is modified: Currently-active LEDs are steadily lit as usual, but in addition, ANY LEDs which have been active since the test was started (or re-started using **[EDIT/CLEAR]**) are flashed.

Entering Event Viewer mode does not interrupt the progress of the test, i.e. further Events are logged irrespective of whether the display is in Event Viewer or Summary mode.

The **[EXIT]** key terminates the logging operation and leaves Watchdog Run to Log mode. Note that an '<END OF TEST>' event is placed in the log at this point in order that the duration of the test is recorded.

After running the Watchdog, the format of the Results Log is:

```
Prism Sound DSA-1 Results Log:
-----

*** BLOCK 01 ***:

[Watchdog:]

001. 001:12:10.0    C-S:CRC Error
002. 001:12:13.4    C-S:CRC OK
003. 002:20:28.8    C-S:Consumer
004. 003:45:48.9    Main Aud Black
005. 003:45:51.1    Main Aud Activ
006. 008:43:55.6    <END OF TEST>
```

It is possible to set up the DSA-1 so that Watchdog Events are accompanied by the transmission of an Event-specific serial code from the RS-232 port. This is done by enabling 'Alarms' in the System Settings sub-menu as described in section 6.9.1.4. If Alarms and Key Beeps are both enabled, each Event is also accompanied by a beep from the DSA-1's loudspeaker.

NOTE: Some LEDs only cause an Event to be logged when they come on , and not when they go off (these are the '**ERROR**', '**N.FAIL**', '**BLOCK**', '**PRO**' and '**CONS**' LEDs); all the others cause Events to be logged both as they are lit and as they are extinguished. A special case is the '**FAIL**' LED which always causes an Event as it lights whether it is alarmed or not, but only causes an Event when it goes out if it is alarmed.

Refer to appendix A for a list of possible Events in Watchdog mode, and their Alarm codes.

6.6.2. Setup (menu 6.2.)

The Watchdog Setup mode is used to select which CARRIER and DATA panel LEDs are to be 'alarmed' prior to running the Watchdog to the Results Log.

On entering Watchdog Setup mode, the LCD shows:

Select alarms on
CARRIER and DATA

The CARRIER and DATA panels no longer display their usual information, but instead show alarmed LEDs lit, and 'masked' (non-alarmed) LEDs extinguished. One LED at a time is selected for adjustment, which flashes. The current LED is selected using the [**<**] and [**>**] keys. The [**EDIT/CLEAR**] key is used to toggle the alarmed state of the current LED, which is indicated by the duty-cycle of its flashing - mostly-on when alarmed and mostly-off when masked.

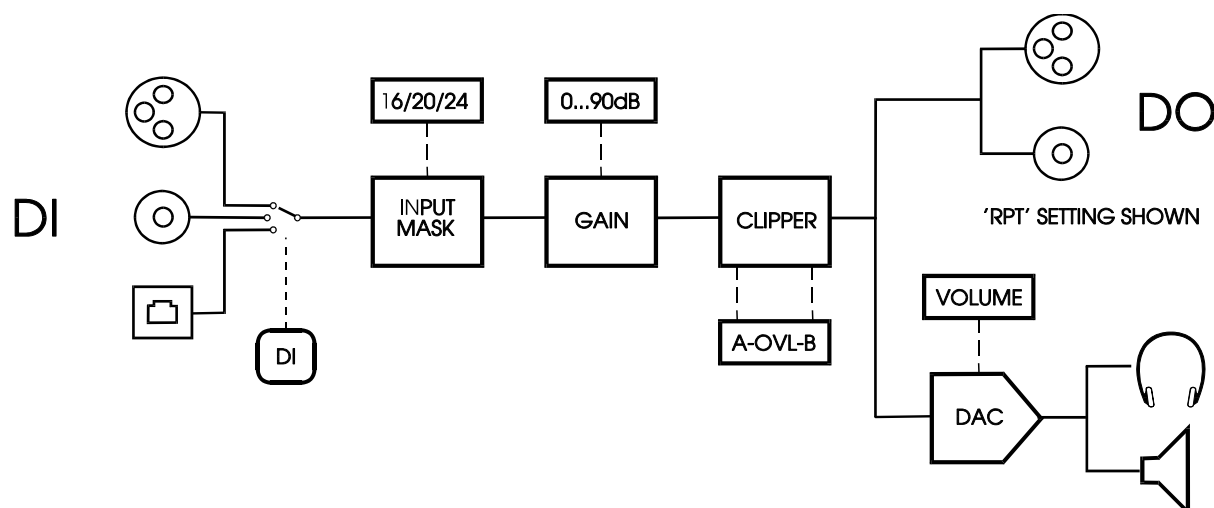
Note that the **'WRT REF'** LED cannot be alarmed for use by the Watchdog.

When the desired pattern of alarms is selected, return to the 'Watchdog' level of the menu system, by pressing [**EXIT**].

6.7. Microscope (menu 7.)

The DSA-1 Microscope mode allows gain to be applied when monitoring a low-level digital audio signal. This is especially useful in evaluating re-dithering or noise-shaping schemes or for testing for possible word-length truncation effects between items of equipment.

A block diagram of the Microscope mode is shown below:



Incoming data from the selected DI connector is first passed through a word-length mask of 16, 20 or 24-bits, which allows the effect of such a truncation to be auditioned. The output of this stage then has digital gain applied (from 0dB to +90dB) before being passed to the DO connector and the DSA-1's monitoring DAC. The **[DO]** key can be used to bypass the processing at the DO connector, allowing 'A/B' insertion of the programmed gain and masking. Separate A and B sub-frame clipping detectors are provided, since this is a particular hazard in the application of high gains. Note that no DC-blocking filter is included, so small amounts of DC at the input can easily cause the Microscope to 'saturate' at high gains. The overload indicators should be monitored for signs of clipping.

The display format for Microscope mode is:

```

→Gain:+12dB
I/P:24b  OVL:
  
```

The amount of applied gain is shown on the upper line, and the input mask-length and clipping indicators are shown on the lower line. The flashing arrow symbol shows whether the gain or input mask-length are currently selected for adjustment.

The **[ENTER/NEXT]** key toggles the current selection between gain and input mask-length, and the **[<]** and **[>]** keys are used to adjust the selected parameters. Clipping

is indicated per sub-frame by the appearance of 'A' and 'B' characters in the lower left-hand corner of the display.

The **[DO]** key should normally be selected to 'RPT' when using the Microscope; selection 'loop-through' mode bypasses the digital processing, connecting DI directly to DO. Note that Microscope mode cannot be entered if the **[DO]** key is set to '**FNC**' for the use of the signal generator.

Pressing the **[EXIT]** key causes the Microscope mode to be discontinued.

6.8. Results Log (menu 8.)

The Results Log menu contains a variety of facilities for examining, clearing and printing the contents of the DSA-1 Results Log.

The DSA-1 Results Log is an 4095-byte area of non-volatile memory which may be filled with various test results, settings etc. Each time some data is placed in the Results Log, it is marked with an 'Item Marker' which describes the source or type of entry. In addition 'Block Markers' can be manually inserted into the Results Log in order to ease organisation of the data. Log data can be printed directly to a serial printer, either as it is being recorded or later. Alternatively, the Log can be uploaded to the Results Log Uploader/Editor software running on a host PC.

A typical Results Log printout might look like this:

```
Prism Sound DSA-1 Results Log:
-----

*** BLOCK 01 ***:

[Carrier Tests:]

XLR input, 110R,
Locked
fs Jitter [P-P]    = 2.9ns fc=700
Data Jittr [P-P]  = 8.5ns fc=700
Amplitude =       3.829 V [P-P]
Closure @ZeroX =  16.6ns, 10.2%
Errors =          0      since last CLEAR
Commonmode Ampl=   677mV [P-P]
Audio-band Ampl=   199mV [P-P]

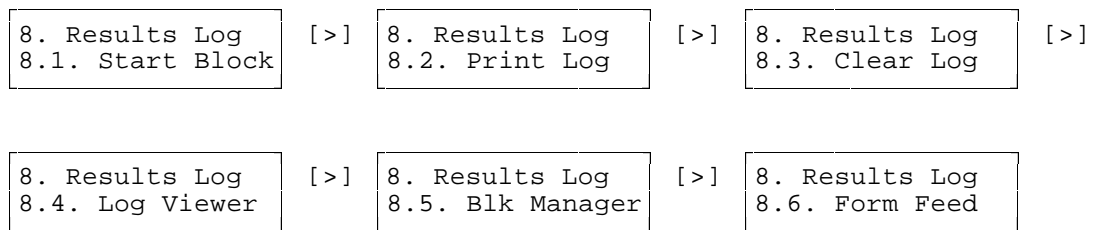
[Test Sequence:]

Strict Pro Test

1. Carrier qual    PASSED
2. Pro CS          PASSED
3. fs frequency    PASSED
4. Amplitude       PASSED
5. Jitter          PASSED
6. Eye width       PASSED
7. Channel Stat    **FAILED CRC error
8. Biphase errs    PASSED
*Tests Complete

*** BLOCK 02 ***:
```

The Results Log menu has six different sub-menus, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

6.8.1. Start Block (menu 8.1.)

Entering the Start Block sub-menu produces a message of the type:

```
Block 2 started  
2418 bytes free
```

A Block Marker is inserted into the Results Log at the beginning of the spare capacity of the Log, after which the menu system automatically returns to the Results Log level.

Alternatively, the following message may be displayed:

```
** Block denied  
2418 bytes free
```

In this case, no Block Marker has been inserted. The reason is usually because no data has been logged since the insertion of the last Block Marker, and therefore the creation of an empty Block is being prevented. Note that this is also the case if you try to insert a Block Marker into a cleared Log, since an empty Log always has a Block 1 Marker at its start. Insertion of a new Block Marker is also prevented if the Log is full or very nearly full.

There are a number of reasons why you might want to insert Block Markers into the Results Log. Firstly, they provide a convenient way of segmenting the Log for ease of later examination. For instance, all tests on a certain day or on a certain item of equipment can be placed in one Block. Note that the Block Manager sub-menu described below allows Log Blocks to be printed or deleted individually, so it makes sense to group tests in this way. Alternatively, if Log Flush mode is enabled (as described in section 6.9.1.8.) then the Log is cleared back to the last Block Marker every time a new Item is to be placed in the Log. This means that each logged Item overwrites the last, unless a Block Marker is inserted after an Item which is required to be kept.

6.8.2. Print Log (menu 8.2.)

Entering this sub-menu causes the entire contents of the DSA-1 Results Log to be printed to a serial printer connected to the RS-232 port. After printing is completed, the menu system automatically returns to the Results Log level. Alternatively, the **[EXIT]** key can be used to halt printing part-way through.

The Block Manager sub-menu described in section 6.8.5 allows individual Blocks of the Results Log to be printed.

The appropriate baud rate for the printer is configured in the Setup menu as described in section 6.9.1.9.

6.8.3. Clear Log (menu 8.3.)

Entering this sub-menu brings up the message:

ENTER to confirm EXIT to cancel

Pressing **[ENTER/NEXT]** causes the Results Log to be cleared, followed by automatic exit to the 'Results Log' level; **[EXIT]** quits directly without clearing the Results Log.

6.8.4. Log Viewer (menu 8.4.)

The Log Viewer allows the Results Log to be examined line by line on the LCD display. The displayed format is prone sometimes to be slightly confusing, since lines of text in the Log are normally longer than the 16 characters of the LCD display. However, the Log Viewer is a quick and easy way to check that the desired Items have been successfully logged.

On entry, the viewing point is the start of the Log. The **[<]** and **[>]** keys scroll up and down through the Log respectively. Pressing **[ENTER/NEXT]** moves the viewing point to the start of the next Block (i.e. to the next Block Marker). If the viewing point is already in the last or only Block, then **[ENTER/NEXT]** moves the viewing point to the end of the Log. The **[EDIT/CLEAR]** key returns the viewing point to the start of the Log.

If key beeps are enabled (see section 6.9.1.1) then a long beep is sounded each time the viewing point lands on a Block Marker. This makes it easy to keep track of the passage of Block Markers when scrolling with the **[<]** and **[>]** keys.

To return to the 'Results Log' level of the menu system, press **[EXIT]**.

6.8.5. Block Manager (menu 8.5.)

The Block Manager allows manipulation of the Results Log, Block by Block. The Block Manager display format is:

Block 2 , 0034 3021 bytes free

The upper line of the display shows the Block number and size of the current Block (initially set to the last Block in the Log, on entering the sub-menu). The lower line shows the spare capacity remaining in the Results Log.

The [**<**] and [**>**] keys respectively scroll up and down the Results Log, Block by Block. The [**EDIT/CLEAR**] key deletes the currently-selected Block from the Log.

The [**ENTER/NEXT**] key causes the current Block to be printed on a serial printer via the RS-232 port. Note that the Log Header (the 'Prism Sound DSA-1 Results Log' message) is only appended to the first Block printed after entering the Block Manager. The appropriate Baud rate for the printer is configured in the Setup menu as described in section 6.9.1.9.

To return to the 'Results Log' level of the menu system, press [**EXIT**].

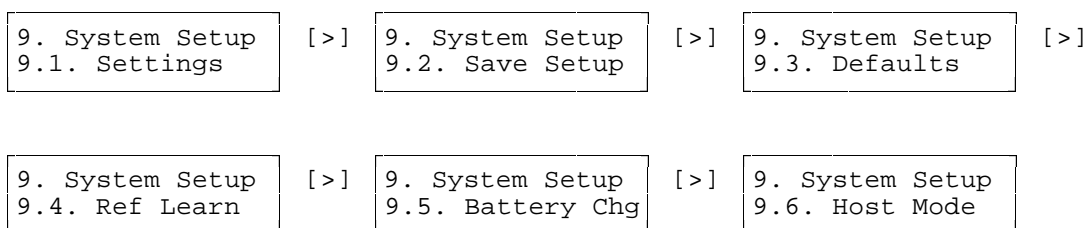
6.8.6. Form Feed (menu 8.6.)

Entering this sub-menu sends a form-feed character (ASCII 12, 0x0C) directly to the serial printer. This is intended to allow convenient ejection of a finished printout. Some printers will require this, some will not.

After sending the form-feed, the DSA-1 automatically returns to the 'Results Log' level of the menu system.

6.9. System Setup (menu 9.)

The System Setup menu deals with a variety of miscellaneous 'housekeeping' functions of the DSA-1. These include various operational mode settings, storage and initialisation of default settings, soft calibration of sampling-frequency measurements, battery charging and host PC uploading/downloading. There are six different sub-menus, shown below, which can be selected using the [**<**] and [**>**] keys.



When the desired sub-menu is displayed, press [**ENTER/NEXT**] to enter that sub-menu. Alternatively, to return to the top level of the menu system, press [**EXIT**].

6.9.1. Settings (menu 9.1.)

The following sections describe the system settings options. There are ten different parameters which can be adjusted, and these are displayed using the [**<**] and [**>**] keys. When the desired parameter is displayed, it is adjusted by using the [**EDIT/CLEAR**] key to cycle through the available options. Press [**EXIT**] to return to the System Setup level.

Note that certain parameters which were adjustable in the Settings menu (4.1.) in version 1.0 of the DSA-1 firmware, such as the jitter corner-frequency and the timecode display mode, are now displayed and adjusted directly within the measurement sub-menus to which they apply.

For each setting, the options can be cycled with the **<EDIT/CLEAR>** key. To return to the System Setup level of the menu system, press **<EXIT>**.

6.9.1.1. Key beeps (menu 9.1. setting 1.)

OPTIONS: OFF, ON

This option enables or disables the audible 'beep' which accompanies a key press or transmission of a Channel Check or Watchdog alarm code.

6.9.1.2. Coax output (menu 9.1. setting 2)

OPTIONS: LOW, HIGH

When the state of the **[DO]** key is 'RPT' or 'FNC' for re-clocking or signal generation, the output carrier amplitude at the BNC DO connector may be selected. The 'HIGH' level voltage is 1V P-P for compatibility with the newly-proposed AES3-ID specification; the 'LOW' level voltage is 500mV P-P for compatibility with consumer SPDIF systems. Note that these voltages assume a correctly terminated line.

6.9.1.3. DAC de-emphasis (menu 9.1. setting 3.)

OPTIONS: AUTO, ON, OFF

The monitoring DAC has a 15/50µs de-emphasis capability which can be set to follow the incoming Channel Status indication ('AUTO'), or can be over-ridden 'ON' or 'OFF'. De-emphasis is applied to the DAC output and therefore ensures the correct audio response for both headphones and the internal loudspeaker.

6.9.1.4. Alarms (menu 9.1. setting 4.)

OPTIONS: OFF, ON

This setting determines whether or not alarm codes will be transmitted from the RS-232 port to mark the occurrence of logged Events in Channel Check and Watchdog modes. When turned ON, the alarms are transmitted as long as the serial port has not been configured for 'live-printing' as described below. If 'key beeps' are also enabled, transmission of the alarm code is accompanied by a beep.

Refer to sections 6.4.9, 6.5 and 6.6 for details of the Channel Check and Watchdog modes, and appendix B for details of the various alarm condition serial codes.

6.9.1.5. Sub-menu memory (menu 9.1. setting 5.)

OPTIONS: ON, OFF

This option causes the last sub-menu selection displayed within each menu to be remembered on exit from the menu so that it is immediately available next time that menu is entered. For example, in the OFF state, entering the Carrier Tests menu would always result in the fs Freq sub-menu (1.1.) being offered, even if a different sub-menu had been in use last time Carrier Tests were performed. In the ON state, if, for example, the Eye Closure sub-menu (1.5.) happened to be on display when the Carrier Tests menu was exited, this would be returned to immediately next time the Carrier Tests menu was entered.

The OFF mode is principally provided for use by the Macro Processor software (see section 7.4.2.) so that menu navigation is more predictable when writing macros. It also provides a similar operation to the version 1.0 firmware.

6.9.1.6. Power save (menu 9.1. setting 6.)

OPTIONS: ON, OFF

In the ON state, this setting causes the DSA-1 to switch itself off if no keys are pressed for a period of five minutes during battery operation. In the OFF state, the DSA-1 will stay on until the battery is exhausted.

6.9.1.7. Live print (menu 9.1. setting 7.)

OPTIONS: OFF, ON

In the ON state, 'live-printing' of the Results Log is enabled. During live-printing, any data entered into the Results Log is also sent to the RS-232 port at the same time.

Operation of the DSA-1 Results Log is described in section 6.8.

6.9.1.8. Log flush (menu 9.1. setting 8.)

OPTIONS: OFF, ON

In the ON state, this setting causes the Results Log to be cleared back to the last Block Marker at the start of any logging operation.

The reason for providing this feature is as follows: In the OFF state, each logging operation gradually adds to the Results Log until it is full. At this point no more logging operations can be carried out until the Log is manually cleared. If Log flush is ON, each logging operation over-writes the previous one in the Results Log, unless the last result is protected by manually inserting a Block Marker. Of course, this does not prevent the Results Log from filling up eventually, but places the onus on the operator to specifically retain any logged results which are of future interest. This option is especially useful in live-printing mode (described in the previous section) where logged results may not need to be kept once they are printed.

Operation of the DSA-1 Results Log is described in section 6.8.

6.9.1.9. Printer baud (menu 9.1. setting 9.)

OPTIONS: 1200, 2400, 4800, 9600

In setting is used to set the Baud rate used when sending Results Log data to the serial printer. Eight data bits are transmitted, without parity and with one stop-bit (xxN81). Neither hardware or software flow-control is exercised, since quantities of print data are quite small and can be comfortably buffered by most printers.

Note that this Baud rate only applies to data intended for a serial printer; other serial transactions such as those with a host PC are always at 9600,N,8,1.

Operation of the DSA-1 Results Log is described in section 6.8.

6.9.1.10. Printer tabs (menu 9.1. setting 10.)

OPTIONS: ON, OFF

During printing or uploading of the Results Log, Block Markers, Item Headings, etc. normally have different levels of horizontal indentation to aid readability of the Log. This setting allows this feature to be turned OFF, for example when using a miniature 40-column printer.

Since miniature printers also usually have only enough buffering for one line of text, the OFF state also causes a short delay to be inserted after transmission of each line.

Operation of the DSA-1 Results Log is described in section 6.8.

6.9.2. Save Setup (menu 9.2.)

Using this sub-menu, preferred settings can be stored as power-up defaults. On entering the sub-menu, you will see the message:

```
ENTER to confirm
EXIT to cancel
```

At this point, pressing the **[EXIT]** key causes a return to the 'System Setup' menu. Pressing **[ENTER/NEXT]** causes the following message to appear:

```
Saving...
```

After a few moments the saving is completed and the DSA-1 displays `OK' and sounds a `beep' on the loudspeaker (if beeps and speaker are enabled). The DSA-1 automatically returns to the `System Setup' menu.

The states of all DSA-1 operating parameters are stored to become the future power-up values.

6.9.3. Defaults (menu 9.3.)

This sub-menu allows the 'factory default' operating parameters to be loaded into the DSA-1. Entering the 'Defaults' sub-menu causes this message to be displayed:

ENTER to confirm
EXIT to cancel

At this point, pressing the **[EXIT]** key causes a return to the 'System Setup' menu. Pressing **[ENTER/NEXT]** causes the default operating parameters to be loaded, and the DSA-1 to restart in the 'Select Mode' (top level) menu. The action of loading the default parameters does NOT cause the power-up values in non-volatile storage to be overwritten; these remain unaltered. If it is desired to initialise the power-up parameters, you must execute sub-menu 9.2. (Save Setup) after instating the defaults.

This facility is often used at the start of macros to enable the Macro Processor software to put the instrument into a known state. This is why it would be undesirable for the action of loading the default parameters to cause loss of the user's power-up values.

The default settings are as follows:

<u>Front-panel controls:</u>	Loudspeaker: OFF
	LS Mode: BOTH
	Volume: 20/31
	DI: XLR
	TERM: Lo-Z
	CHAN: A
	DO: Loop-thru
	SYNC: CHAN

<u>Carrier Tests:</u>	Jitter corner frequency: 1.5kHz
	Phase units: Degrees
	Jitter units: ns
	Eye threshold: Zero-crossing

<u>Data Tests:</u>	Bargraph mode: No numeric readout, no hold, mono
	C-S timecodes: Sample-count

<u>Test Sequences:</u>	Current Sequence: Sys01, Global
------------------------	---------------------------------

<u>Generator/Editor:</u>	Generator mode: GEN1, Internal Sync Source, 48kHz Edit Mode: TIED, Aud:Y, C-S:Y, V:Y GEN1 Audio: Sine, -14dBFS, 1kHz, ON, 24b GEN1 C-S: Field 1 edited to Pro, rest transparent GEN2 Audio: fs/4 sine, ON GEN2 C-S: Professional Valid bit: 0, Valid Jitter generator: PRS-H, OFF
<u>Channel Check:</u>	Settings: Both channels, 24b
<u>Watchdog:</u>	Settings: All LEDs masked
<u>Microscope:</u>	Settings: 0dB, 24b
<u>Settings:</u>	Key beeps: OFF Coaxial O/P amplitude: LOW DAC de-emphasis: AUTO Alarms: OFF Sub-menu memory: ON Power save: ON Live print: OFF Log flush: OFF Printer baud: 1200 Printer tabs: ON

6.9.4. Ref Learn (menu 9.4.)

This feature provides a means of software-calibrating the unit against an external reference signal, and storing the calibration so that sampling frequency accuracy measurements may be made with respect to the reference, even after it has been disconnected. In this state, the DSA-1 is said to have 'learned' the external reference signal.

Having entered 'Ref Learn' mode, the LCD displays:

ENTER to acquire Ref fs, or CLEAR

At the same time, the DSA-1 automatically selects whichever REF SYNC input is receiving the reference, and determines whether it is in IEC958/AES11 or Wordclock format. The CARRIER panel indicates the sampling rate and accuracy of the reference. At this point, the [<] or [>] key will toggle the LCD display between the message shown above, and a readout of the measured sampling frequency and

accuracy as described in section 6.1.1. Note, however, that in this case the measurement applies to the REF SYNC input rather than DI.

When the reference is being satisfactorily recognised, pressing the **[ENTER/NEXT]** key causes the reference to be learned, and all subsequent sampling rate measurements will be made relative to the external REF SYNC signal even after it has been disconnected. This state is indicated by the **'WRT REF'** indicator being lit in the CARRIER panel and an asterisk being displayed in the bottom right-hand corner of sub-menu 1.1 (fs freq).

Alternatively, if a return to measurement with respect to the DSA-1's internal reference is desired press **[EDIT/CLEAR]** to clear the calibration store.

If neither re-calibration nor clearing is required, press **[EXIT]** to return to the 'Settings' menu.

For optimum precision, it is recommended that Ref Learn be used after the DSA-1 has been switched on for a few minutes.

NOTE: The DSA-1 can be used measure the frequency of a Wordclock input by connecting the Wordclock to the coaxial REF SYNC input, entering the 'Ref Learn' sub-menu and selecting the fs display using **[<]** or **[>]**. You can exit without 'learning' the Wordclock by simply pressing **[EXIT]**.

6.9.5. Battery Charge (menu 9.5.)

Whenever the DSA-1 is operated using the DC adaptor, the battery is trickle-charged. However, it would take a long time to fully charge a low or exhausted battery in this way.

The 'Battery Charge' sub-menu allows the DSA-1's battery to be fully discharged and then fast-charged from the DC adaptor. The DSA-1 cannot be operated during fast battery charging.

On entering the 'Battery Charge' sub menu, the following message is shown after a few moments:

Battery charging
(discharge)

While the battery is discharging, all the LEDs on the CARRIER and DATA panels are lit and the DSA-1 controls are frozen. When the battery is fully discharged, the display changes to:

Battery charging
(12 hr rate)

After this, the battery is charged for 12 hours and then the DSA-1 switches off.

It is possible to curtail the charging process at any time during the discharging or charging cycle and to return to normal operation by pressing **[EXIT]**. The DSA-1 can also be manually turned off in the middle of a charging operation if required.

Whilst it is preferable to use the full discharge/charge cycle in order to retain the maximum capacity of the battery, it is possible to manually jump straight into the charging part of the cycle by pressing **[ENTER/NEXT]** during discharge. This can save time if urgent topping-up of a low battery is required. The DSA-1's battery contains high-temperature sintered NiCd cells which are designed to withstand some amount of overcharging.

6.9.6. Host Mode (menu 9.6.)

It is necessary to enter Host Mode for uploading Results Log data to a host PC, or for downloading new User Test Sequences via the RS-232 port.

On entering Host Mode the following message is displayed:

Host Mode
idle

When the Host PC is transacting data the 'idle' message is briefly replaced by a message describing whatever type of transfer is taking place.

Operation of the DSA-1 in conjunction with a Host PC is described in section 7.

To return to the 'System Setup' menu, press **[EXIT]**.

7. Using the DSA-1 with a host PC

The serial communications port of the DSA-1 allows for a variety of extended modes of operation. This section deals with some of the ways you can use the DSA-1 with a host PC, and describes the 'DSA-1 Host Support' suite of software supplied with the DSA-1.

The DSA-1 Host Support software consists of:

MS-Windows applications:	Results Log Uploader/Editor Test Sequence Compiler/Downloader
MS-DOS applications:	Macro Processor Firmware Downloader

7.1. Installing the DSA-1 Host Support software

The DSA-1 firmware and Host Support software (and its ancillary files) are provided on a floppy diskette; a backup copy of this diskette should be made for safe-keeping. Any subsequent updates to the DSA-1 firmware or Host Support software will be issued on a similar diskette.

Two installation procedures need to be performed, one for the MS-Windows applications and another for the MS-DOS applications. If one or other set is not required, then the associated installation procedure need not be performed. For example, if the Firmware Downloader and Macro Processor are not needed, then the MS-DOS installation can be skipped.

7.1.1. MS-Windows software installation

Place the DSA-1 Host Support diskette in drive A: (for example) and select FILES:RUN from the MS-Windows Program Manager menu bar. Enter

A:\SETUP.EXE

as the file to be run, and click 'OK'. Installation is then conducted automatically, prompting the user for any settings required, such as the preferred installation target directory name.

After installation, a DSA-1 Host Support program group is automatically created from which the applications are run.

7.1.2. MS-DOS software installation

Place the DSA-1 Host Support diskette in drive A: (for example) and, at the MS-DOS prompt type:

A:\INSTALL<Enter>

Installation is then conducted automatically, prompting the user for any settings required, such as the preferred installation target directory name.

The default target directory name for the MS-DOS software is c:\DSADOS. You should run the MS-DOS applications from within this directory (or whichever target directory you have specified).

7.2. Results Log Uploader/Editor software

7.2.1. Introduction

This application is designed to allow the operator to upload the DSA-1 Results Log (as described in section 6.8) to a host PC. It is then possible to print the Log or import it into documents or reports.

The Results Log Uploader/Editor is an MS-Windows application which requires MS-Windows version 3.1 or higher.

7.2.2. Using the Results Log Uploader/Editor

7.2.2.1. Starting the program

The DSA-1 Results Log Uploader/Editor is started by double-clicking the appropriate icon in the 'DSA-1 Host Support' program group.

A large editor window appears which is similar to most basic MS-Windows text editors, such as Notepad. The FILE menu contains an additional Upload option which allows the Results Log to be loaded serially into the editor from the DSA-1. Familiar menu options then allow editing, printing, copying etc. in the usual way.

The following sections document the functions of the pull-down menus.

7.2.2.2. FILE menu

The FILE menu contains the following options:

Upload Uploads the Results Log from a DSA-1 for editing

On selecting Upload, a dialog box appears prompting the operator to start the upload. It is necessary to ensure that the DSA-1 is connected to a COM port of the PC with a null-modem cable as described in section 3.6, and that it is in Host Mode (sub-menu 9.6.). The COM port can be selected in the dialog box; this setting is preserved in an initialisation file on disk, and so is retained indefinitely until it is changed.

On starting the operation, messages on the DSA-1 display and in the PC dialog box show that the transfer is taking place. If anything is wrong, a message box describing the error is displayed on the PC screen. If all goes well, the Results Log appears in the editor window.

New	Open a new, empty text file for editing
Open	Open an existing text file for editing
Save	Save the currently active file
Save As	Save the currently active file using a new filename
Print	Print the currently active file
Print Setup	Select the printer options for the above
Exit	End the program

7.2.2.3. EDIT menu

The EDIT menu contains the following options. Some may be 'greyed' if not available.

Undo	Undo the last editor action
Cut	Cut marked text to the clipboard
Copy	Copy marked text to the clipboard
Paste	Paste text from the clipboard at the cursor
Delete	Delete marked text
Date/Time Stamp	Insert the current date and time at the cursor

7.2.2.4. SEARCH menu

The SEARCH menu contains the following options:

Find	Search for specific text
Replace	Search and replace specific text
Next	Perform the next search or replace action

7.2.2.5. WINDOW menu

The WINDOW menu contains the following options:

Tile	Tile the editor windows
Cascade	Cascade the editor windows
Arrange Icons	Rearrange any icons representing minimized editor windows
Close All	Close all editor windows
Save state	Save the current state of the editor desktop to a disk file
Restore state	Restore the current state of the editor from the above disk file

7.2.2.6. ABOUT menu

Clicking on the ABOUT pull-down causes a message box to be displayed showing the release details of the program.

7.3. Test Sequence Compiler/Downloader software

7.3.1. Introduction

This application is designed to allow the user to create his own Test Sequences which are then serially loaded into the User Test Sequence stores of the DSA-1, which are described in section 6.3. This is done by creating 'source files' using the built-in text editor, then 'compiling' these to produce binary files. The binary files are then downloaded into the appropriate stores in the DSA-1. All of these steps are handled by the same application. Source files corresponding to the four built-in 'System' Test Sequences are provided, and it is useful to use these as a starting point when writing User Sequences for the first time.

The Test Sequence Compiler/Downloader is an MS-Windows application which requires MS-Windows version 3.1 or higher.

7.3.2. Elements of a Test Sequence

This section gives a brief overview of the structure of a DSA-1 Test Sequence. It is recommended that the reader refer to the listings of the built-in sequences in appendix C which provide a useful example.

The idea of a Test Sequence is that an operator can select and run a particular set of tests on an input carrier, and be informed that these have all been accomplished successfully. This gives a high degree of confidence in a very short time. Any failure is accompanied by an informative message, after which the whole Sequence can be restarted, or the offending part can be repeated, or that part can be skipped and the remainder of the Sequence can be run.

Thus Test Sequences are built up from one or more 'Segments'. Each Segment may run successfully or may fail. The name and number of each Segment is displayed on the DSA-1's LCD while it is running, and accompanies any failure message sent from that Segment. The Segment boundaries are also the only point at which the Sequence can be restarted if partial execution is desired.

The body of each Segment is made up of a number of 'Instructions', 'Labels' and 'Comments'. Each Segment has a Start Instruction (which includes the name of the Segment), and one or more End Instructions. A PASS/FAIL flag must be appropriately set before executing the Segment End instruction, in order to govern whether the Sequence will stop and display a message (FAIL) or carry on to the next Segment (PASS).

The following example shows a very simple Segment called 'Carrier lock' which FAILS if the selected DI is unlocked, displaying the message 'No carrier lock'; otherwise the Segment PASSES. It would be unusual to write a Segment as simple as this, since normally a Segment contains a number of related tests rather than just one. The operation of the example Segment should be clear to anyone who has written a computer program in almost any language:

```
/* Segment 1: tests that carrier is locked */
SEGMENT( 1, 'Carrier lock' );
SET_FAIL;
TEST_UNLOCK_BIT; JUMP_IF_ZERO(@lock_ok);
MESSAGE( 'No carrier lock ' ); SEGMENT_END;
@lock_ok:
SET_PASS;
SEGMENT_END;
/*****/
```

Instructions may appear on lines on their own, or with several on one line separated by semicolons (';') as in the example above. Instructions may be 'simple', with no parameters, or they may be 'complex' with following parameters enclosed in brackets:

SET_PASS simple Instruction

MEAS_FS_JITTER(6) complex Instruction

Complex Instructions are followed by one or more parameters in brackets, separated by commas. Parameters may be:

- Numeric (binary, base 10 or hexadecimal);
- Text (in single quotes);
- A Label (without the final colon);

Labels are typically targets for branching ('jump') Instructions. A Label is distinguished from an Instruction by starting with the '@' character and ending with a colon (':'), for example:

@MYLABEL:

Labels may be on a line on their own, or may precede one or more Instructions on the same line.

Comments may be added to the segment definitions in two ways; enclosed within '/' and '*' delimiters or with the '//' form which defines a Comment that lasts to the end of a line:

```
/* This is a comment */      JUMP(@start)

// So is this, anything on this line is ignored from here
```

Comment text is ignored by the compiler and serves only to make a Test Sequence source file easier to read.

7.3.3. Test Sequence Instruction reference

7.3.3.1. Sequence Start and End

A **HEADER** Instruction **MUST** be the first Instruction in the source file. It defines the start of the Test Sequence, and includes a 16 character text message that appears on the DSA-1 display when the Sequence is selected or run, for example:

```
HEADER('Special Test      ')
```

Similarly, the final Instruction of the Sequence should be:

```
END
```

7.3.3.2. Segment Start and End

These Instructions define the beginning and end of a Segment (the concept of a Test Sequence Segment is described above). Here is an example of a Segment start Instruction; the first parameter defines the running-order position of the Segment, the second is a message to be displayed when the segment runs or fails:

```
SEGMENT( 3, 'fs Tests      ')
```

The Segment end instruction has no parameters and causes the DSA-1 to skip to the beginning of the next segment, or to end the sequence if there are no more Segments (or if the Segment has failed):

```
SEGMENT_END
```


7.3.3.3. Set switches

These Instructions have no parameters and are used to set the DSA-1 switched operating parameters:

SELECT_COAX SELECT_XLR SELECT_OPT	Digital Input (DI) selection
SELECT_CHANA SELECT_CHANB	Sub-frame (channel) selection
SELECT_JITTERCF_HI SELECT_JITTERCF_LO	Jitter PLL corner-frequency selection
SELECT_EYE_AT_ZEROX SELECT_EYE_AT_200MV	Threshold voltage for eye-narrowing measurement
SELECT_LOZ SELECT_HIZ	Select DI termination state
CLEAR_ERROR_COUNT	Clear the biphas/parity error counter
CLEAR_LOG	Clear the Results Log
INSERT_LOG_BLOCK	Insert a Block Marker into the Results Log
CLEAR_LOG_LAST_BLOCK	Clear the Results Log back to last Block Marker

7.3.3.4. Test binary results

These Instructions cause a snapshot of a particular DSA-1 operating parameter to be loaded into the 16-bit Test Register. For most of these Instructions, the value is only a single bit, which is loaded into the least-significant bit of the Test Register with the upper 15 bits zeroed. Where indicated, a longer value is loaded at the least-significant end of the Test Register with the remaining upper bits zeroed.

TEST_UNLOCK_BIT	CARRIER panel 'FAIL' LED; 1 if lit
TEST_NFAIL_BIT	CARRIER panel 'N.FAIL' LED; 1 if lit
TEST_BLOCKERR_BIT	CARRIER panel 'BLOCK' LED; 1 if lit
TEST ALOG_BIT	CARRIER panel 'ALOG' LED; 1 if lit
TEST_AUX_BITS	DATA panel 'AUX' LED; 1 if lit
TEST_AUDL_BITS	DATA panel 'AUDIO' (low-bits) LED; 1 if lit
TEST_AUDMAIN_BITS	DATA panel 'AUDIO' (main bits) LED; 1 if lit
TEST_V_BIT	DATA panel 'V' LED; 1 if lit
TEST_U_BIT	DATA panel 'U' LED; 1 if lit
TEST_ANEB_BIT	DATA panel 'A≠B' LED; 1 if lit
TEST_CRCERR_BIT	DATA panel 'CRC ERROR' LED; 1 if lit

TEST_COAX	1 if [DI] is set to 'COAX'
TEST_XLR	1 if [DI] is set to 'XLR'
TEST_OPT	1 if [DI] is set to 'OPT'
TEST_LOZ	1 if [TERM] is set to 'LO-Z'
TEST_HI_Z	1 if [TERM] is set to 'HI-Z'
TEST_CHANA	1 if [CHAN] is set to 'A'
TEST_CHANB	1 if [CHAN] is set to 'B'
TEST_JITTERCF_HI	1 if jitter PLL corner-frequency is low
TEST_JITTERCF_LO	1 if jitter PLL corner frequency is high
TEST_EYE_AT_ZEROX	1 if threshold for eye-narrowing is at zero-crossing
TEST_EYE_AT_200MV	1 if threshold for eye-narrowing is at 200mV
TEST_LOG_FULL	1 if Results Log is full
TEST_REF_STATUS	(8-bit) loads Ref_Status register: Bit1, 1 if Ref AES11 Rx is unlocked Bit3, 1 if Ref Coax input is active Bit4, 1 if Ref XLR input is active Remaining bits undefined
TEST_CSBYTE(n)	(8-bit) loads incoming Channel Status byte n (n=0..23)

7.3.3.5. Measure numeric results

These Instructions cause a test to be performed using the specified DSA-1 function, after which the numeric result is loaded into the Test Register in the units indicated. Some of the Instructions require that a 'time' parameter is specified which determines how long the measurement is averaged before loading the result.

MEAS_ERROR_COUNT	Biphase/parity error count
MEAS_FS_JITTER(time)	fs jitter (ns)
MEAS_DATA_JITTER(time)	Data jitter (ns)
MEAS_EYE_NARROW(time)	Eye-narrowing (ns)
MEAS_DIF_AMPL	Differential input amplitude (V)
MEAS_CM_AMPL	Common-mode input amplitude (V)
MEAS_LOBAND_AMPL	Audio band input level (V)
MEAS_FS_BASE	Base fs of DI input returns 0 (32k), 1 (44k056), 2 (44k1), or 3 (48k)
MEAS_FS_ERROR	fs error from the standard rate (ppm)
MEAS_PHASE	Carrier phase versus Ref Sync (degrees)

Those Instructions which take a measurement duration as a parameter use units of seconds if the 's' suffix is specified, or tenths of a second if not, e.g.:

```
MEAS_DATA_JITTER( 4.5s )
```

or

MEAS_DATA_JITTER (45)

would both measure Data Jitter over a period of 4.5 seconds.

7.3.3.6. Boolean operators

These Instructions cause a boolean operation to be performed on the Test Register, and are used to operate on the result of a previous TEST_.. Instruction (normally TEST_REF_STATUS or TEST_CSBYTE(n), see above). Conditional branch Instructions may then be used to vary action according to the results of the boolean operation.

AND(value)	Bitwise AND operation: new result = old result AND value
OR(value)	Bitwise OR operation: new result = old result OR value
XOR(value)	Bitwise XOR operation: new result = old result XOR value

'value' is a 16-bit word value expressed either as hexadecimal with a following 'h', binary with a following 'b' or decimal without either. 'value' is enclosed in brackets.

7.3.3.7. Sequence flow

These Instructions cause the DSA-1 to branch or 'jump' to another part of the Sequence. Care should normally be taken **never to branch outside the current Segment**; although this is not specifically prevented, it is only occasionally useful and, if used unwisely, can cause unpredictable results.

The conditional branch Instructions enable different responses to occur depending on the results of previous TEST_.. or MEAS_.. operations. All JUMP_.. Instructions have a label description (starting with a '@' character) as their last parameter. Some also have a hexadecimal, decimal or binary value as the first parameter. Note that all the JUMP_.. Instructions have a 'short form' as shown in the list below.

Conditional branch Instructions act upon the current contents of the Test Register but do not change it, so that multiple JUMP_.. instructions may be used to test the same result several times.

JUMP(@label) or JMP(@label)	Unconditional
JUMP_IF_EQUAL(value, @label) or JE(..	if result = value
JUMP_IF_NOT_EQUAL(value, @label) or JNE(..	if result \neq value
JUMP_IF_GREATER_THAN(value, @label) or JG(..	if result > value
JUMP_IF_LESS_THAN(value, @label) or JL(..	if result < value
JUMP_IF_GREATER_OR_EQUAL(value, @label) or JGE(..	if result \geq value
JUMP_IF_LESS_OR_EQUAL(value, @label) or JLE(..	if result \leq value
JUMP_IF_ZERO(@label) or JZ(..	if result = 0
JUMP_IF_NOT_ZERO(@label) or JNZ(..	if result \neq 0
JUMP_IF_MASKED(value, @label) or JMASKZ(..	if (result AND value) = 0
JUMP_IF_NOT_MASKED(value, @label) or JMASKNZ(..	if (result AND value) \neq 0

The @label parameter must begin with a '@' character. Note that labels when referred to in a JUMP_.. instruction do not have a following colon (':') character. Jumps may be made backwards or forwards.

The JMASKZ and JMASKNZ instructions are most useful for testing Channel Status fields.

Notes on the 'value' parameter:

The value parameter can be expressed in a variety of units depending on what type of result is being tested. Usually there is a 'user-friendly' unit which requires the use of a suffix. Omitting the suffix causes the DSA-1's internal 'raw' units to be used:

Time (MEAS_FS_JITTER, MEAS_DATA_JITTER, MEAS_EYE_NARROW)

Use units of nanoseconds, with the suffix 'ns';

Raw DSA-1 units are multiples of 1/64th of a nanosecond;

```
JUMP_IF_GREATER( 6.5ns, @bad_jitter )
```

is equivalent to

```
JUMP_IF_GREATER( 4160, @bad_jitter )
```

Amplitude (MEAS_DIF_AMPL, MEAS_CM_AMPL, MEAS_LOBAND_AMPL)

Use units of Volts or millivolts, with suffixes of 'V' or 'mV';

Raw DSA-1 units are multiples of 10 mV;

```
JUMP_IF_LESS_THAN( 240mV, @no_level)
```

is equivalent to

```
JUMP_IF_LESS_THAN( 24, @no_level)
```

Frequency (MEAS_FS_ERROR)

Use units of parts per million depending on the sampling frequency in use, with suffixes:

```
'PPM_480' if fs=48.0kHz
```

```
'PPM_441' if fs=44.1kHz
```

```
'PPM_440' if fs=44.056kHz
```

```
'PPM_320' if fs=32.0kHz
```

Raw DSA-1 units are multiples of 1/64 Hz;

Phase (MEAS_PHASE)

Use units of degrees with a suffix of 'DEG';

Raw DSA-1 units are multiples of 360/512 degrees (about 0.703 degrees);

```
JUMP_IF_GREATER_THAN( 45deg, @notinphase)
```

is equivalent to

```
JUMP_IF_GREATER_THAN( 64, @notinphase)
```

7.3.3.8. Segment pass/fail operators

These Instructions set and act upon the PASS/FAIL flag. If set to FAIL before a Segment end, this flag causes the Sequence to abort. If set to PASS before a Segment end, the flag allows the sequence to continue.

SET_PASS	Set flag to 'PASS'
SET_FAIL	Set flag to 'FAIL'
JUMP_IF_PASS(@label) or JPASS(. .	Jump if flag is set to 'PASS'
JUMP_IF_FAIL(@label) or JFAIL(. .	Jump if flag is set to 'FAIL'

7.3.3.9. Miscellaneous

This Instruction displays a message on the bottom line of the DSA-1's display. The text parameter must be enclosed in single quotes:

MESSAGE(text) Show the text on row 2 of the display.

The no-operation Instruction is:

NOP

This causes the DSA-1 to take no action and pass on to the next Instruction.

Long delays are accomplished with:

WAIT(delay)

The delay parameter can be expressed in seconds by using an 's' suffix, otherwise is assumed to be expressed in tenths of a second. There are a number of circumstances where delays need to be inserted, for example waiting for lock to be acquired after changing the [DI] selection.

7.3.4. Using the Test Sequence Compiler/Downloader

7.3.4.1. Starting the program

The DSA-1 Test Sequence Compiler/Downloader is started by double-clicking the appropriate icon in the 'DSA-1 Host Support' program group.

A large editor window appears which is similar to any basic MS-Windows text editor such as Notepad. Familiar menus allow Test Sequence source files to be created, saved, loaded, printed etc. The COMPILE menu contains the additional options for compilation of the source files and downloading of the resulting binary files to the DSA-1.

Try loading, modifying, compiling and downloading one of the standard Test Sequence source files supplied (if you save the result, use 'save as' so as not to overwrite the original standard file). This will give you a feel for the process.

The following sections document the functions of the pull-down menus.

7.3.4.2. FILE menu

The FILE menu contains the following options:

New	Open a new, empty text file for editing
Open	Open an existing text file for editing
Save	Save the currently active file
Save As	Save the currently active file using a new filename
Print	Print the currently active file
Print Setup	Select the printer options for the above
Exit	End the program

Text (source) files have the '.TXT' extension by default, but any other extension except '.BIN' and '.LST' may be used.

7.3.4.3. EDIT menu

The EDIT menu has the following options. Some may be 'greyed' if unavailable:

Undo	Undo the last editor action
Cut	Cut marked text to the clipboard
Copy	Copy marked text to the clipboard
Paste	Paste text from the clipboard into the editor
Delete	Delete marked text

7.3.4.4. SEARCH menu

The SEARCH menu contains the following options:

Find	Search for specific text
Replace	Search and replace specific text
Next	Perform the next search or replace action

7.3.4.5. COMPILE menu

The COMPILE menu contains the following options:

Compile Compile the currently active file into a binary Test Sequence

This option brings up a dialog box containing the text 'Ready to compile...' and a single 'Go' button. When pressed, the 'Go' button starts the compilation process. A count of the lines compiled is displayed. Depending on the result of the compilation, one of two actions is taken:

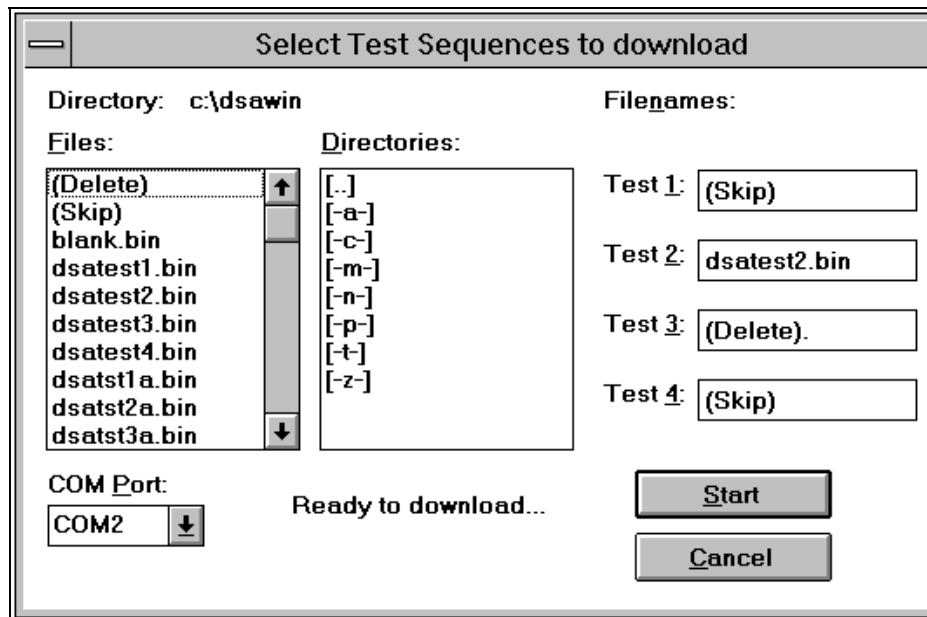
If the compilation succeeds, a file is written containing binary data representing the compiled Test Sequence. This file has the same root name as the source file but has the extension '.BIN'.

If the compilation fails, no binary output file is written.

In both cases a '.LST' list file is created which contains details of the compilation. This file is automatically loaded into the text editor if the compilation fails. Consult section 7.3.4.8 for details of the format of the list file.

Download Download up to four User Test Sequences to the DSA-1

This option brings up a large dialog box containing a filename/directory listing, four edit boxes containing the '*.bin' wildcard filenames, a box containing the current COM port, and two buttons labelled 'Start' and 'Cancel':



The four boxes on the right correspond to the four User Test Sequence stores in the DSA-1. To prepare for a download, the binary filenames to be downloaded are placed in the appropriate boxes. This is done by first clicking in the box to be filled, then on the filename to be placed in it (alternatively this can be done from the keyboard in the normal fashion). The '(Delete)' and '(Skip)' options are entered instead of filenames to allow DSA-1 stores to be erased or left unchanged respectively during the subsequent download operation.

The COM port selection is changed by clicking on the button to the right of the displayed port name, which causes a selection list to appear. Any of COM1..COM4 may be selected. This setting is stored in an initialisation file on disk so that it is retained indefinitely until changed.

When the desired batch of filenames have been loaded into the appropriate boxes, and the correct COM port chosen, the download can be started by clicking on the 'Start' button. The DSA-1 must be connected to the selected COM port with a null-modem cable (see section 3.6), and Host Mode (sub-menu 9.6.) must be selected; if this is not the case, an error message box is displayed. The DSA-1 displays a message in the bottom row of the LCD display to show that downloading is taking place, and a similar message appears on the PC in the dialog box. When the process is complete, the PC shows a message box to that effect, and the DSA-1's **[EXIT]** key can be used to quit Host Mode, at which point a 'Saving...' message appears briefly as any updated Sequences are stored in non-volatile memory. These Sequences are immediately available for selection in Test Sequence Select sub-menu 3.2.

7.3.4.6. WINDOW menu

The WINDOW menu contains the following options:

Tile	Tile the editor windows
Cascade	Cascade the editor windows
Arrange Icons	Rearrange any icons representing minimized editor windows
Close All	Close all editor windows
Save state	Save the current state of the editor desktop to a disk file
Restore state	Restore the current state of the editor from the above disk file

7.3.4.7. ABOUT menu

Clicking on the ABOUT pull-down causes a message box to be displayed showing the release details of the program.

7.3.4.8. List File format

A List File is generated by the compiler during every compilation and contains details of the compiler output and any errors found. The format of the file is as follows for each source-line compiled:

```
[error messages if any]
Line number:  Source file text for this line
               Binary output offset: Binary output bytes  (the compiled opcodes)
```

Thus for a typical line:

```
040:  MEAS_FS_ERROR; AND(07FFFh);
00ba:  38 3b 7f ff
```

Error messages are detailed in a following section and may contain part of the offending line or Instruction.

At the end of the List File a list of all the jumps made is shown, together with any errors caused by missing or unresolved labels. The file ends with a message showing how many lines were compiled in total.

List Files may be loaded into the text editor using the FILE:OPEN menu option. They are automatically loaded if any compilation errors occur.

7.3.4.9. Compiler error messages

The following error messages may appear in the List File:

Not enough memory!

The system cannot start the compiler as not enough memory is available. Close some active windows and try again.

Data error writing output file xxxxx

A disk error was encountered writing the .BIN file.

Data error reading input file xxxxx

A disk error was encountered reading the input file; for example it is not a true plain DOS text file, or does not exist.

Can't open list file xxxxx - disk full?

A list file cannot be written, which could be due to the disk being too full.

Incomplete opcode :

The Instruction following has too few parameters specified. Refer to the Test Sequence language reference.

Bad opcode :

The Instruction following has not been recognized. Refer to the Test Sequence language reference.

Too many JUMPs (n is maximum)

There are too many JUMP Instructions in the test sequence. Rewrite the sequence and try again. The maximum 'n' is currently 100.

Maximum sequence length exceeded!

The compiled length of the Test Sequences must, when compiled, be less than 992 bytes. Reduce the length of the sequence and try again.

Line too long :

The line following is too long. Break the line up and try again.

Sequence must start with HEADER statement!

Every Test Sequence must have a HEADER Instruction at the beginning. Refer to the DSA-1 Test Sequence Instruction reference.

Message string too long!

The required length for the text-string element of a HEADER and MESSAGE Instruction is 16 characters. For a SEGMENT Instruction the required length is 12 characters. The above error message is displayed if more than the required number of characters are entered in the Instruction. If too few characters are entered, no error message results and the string is end-padded with spaces.

7.4. Automated and Remote operation

NOTE: The DSA-1 RS-232 port has the same pinout as a PC-AT COM port, so a null-modem cable (see section 3.6) should be used for Serial Slave operation (if the controlling device is an IBM-compatible PC), and for Macro Processor operation.

7.4.1. Serial Slave mode

Serial Slave mode allows the DSA-1 to be remote-controlled via its RS-232 port. This means that automatic test systems can use a DSA-1 without operator intervention. Writing a controller application program to use a DSA-1 in Serial Slave mode is a difficult programming task and is not recommended to the novice. To conduct automatic tests using the DSA-1 under control of a host PC, the Macro Processor software described below makes the job very much easier.

In Serial Slave mode, the DSA-1 accepts 'key strokes' from the serial input as well as from the keypad, and echoes all LED and display information to the serial output. Serial Slave mode may be invoked and terminated by the external device (typically a host PC) sending special character strings to the DSA-1. Note that this cannot occur when the DSA-1 is in Host Mode (sub-menu 9.5.), nor should Host Mode be entered when Serial Slave mode is active.

For experienced programmers, the serial protocols for entering and operating Serial Slave mode are covered in appendix B.

7.4.2. Macro Processor software

7.4.2.1. Introduction

The Macro Processor is an MS-DOS application which is used to operate the DSA-1 automatically. The Macro Processor invokes Serial Slave mode in the DSA-1, then sends sequences of DSA-1 key-press tokens to the unit to control its actions. Results from the DSA-1's display and LEDs can be read and processed, and output can be sent to printer, file or serial port (other than the port to which the DSA-1 is connected).

The Macro Processor operates on a Macro File, which is prepared by the user. This is a simple ASCII text file containing the instructions ('Keywords') used to control the DSA-1 and act on its results. Macro Files can be prepared using any simple text editor, such as MS-DOS 'EDIT' or MS-Windows 'NOTEPAD'.

7.4.2.2. Macro File format

Macro Files may contain the following:

- Keywords - the various instructions that make up the Macro;
- Labels - usually the destination of JUMP instructions;
- Comments - remarks to aid readability, ignored by Macro Processor;

Keywords are the individual commands to the Macro Processor. These may be separated by semicolons or spaces on the same line to reduce the file length if required. Each Keyword is processed in sequence and causes an effect either within the DSA-1 or in the host PC. JUMP -type Keywords enable loops and other program flow changes.

Labels are usually the destinations of JUMP-type Keywords. They may be up to 20 characters long and always start with an '@' character. The final colon is used at the label point but NOT in the reference to it. After branching, execution of Macro File resumes at the Keyword immediately following the destination label. Labels take the form:

```
JUMP(@bad_jitter)           // reference to label

/* here is the label itself... */
@bad_jitter:  BEEP
```

Comments exist in two forms, both illustrated in the label example above. The `/**` form causes all further text on that line to be considered a comment and thus be ignored by the Macro Processor. The `/*...*/` form encloses the comment (which may be on several lines) but allows active Keywords to follow the final `*/` on the same line. Comments have no effect on the action or flow of the Macro File, and merely serve to explain the Macro File's actions to increase readability.

The Macro File extract below enters the fs frequency sub-menu (the extract assumes that the menu system is offering sub-menu 1.1. on entry), records the result and takes various actions according to the accuracy of the fs reading:

```
ENTER                      // enter fs frequency sub-menu
GET_DSA_VAL(1,6,13);       // get fs (5 digits + two decimals)
PRINT("\n\r fs is \v.");   // and print it out

/* Now test the fs value - fail if more than 1Hz from 48kHz */
JL(47999,@bad); JG(48001.0,@bad) // fs is out of range
PRINT("fs PASSED");        // OK
JUMP(@cont);               // and continue

@bad:
PRINT(" *** fs outside 47999 - 48001 Hz"); // fail message
WAIT_ANY_KEY
```

Two example macro files (EXAMPLE1.MAC and EXAMPLE2.MAC) are loaded into the MS-DOS installation directory along with the DSAMAC.EXE program. These show some typical Macro Processor techniques.

7.4.2.3. Keyword reference

The following is a list of all available Keywords and their actions:

7.4.2.3.1. DSA-1 key presses

These Keywords cause the DSA-1 to react as if the corresponding key had been pressed. This allows the user to navigate the menu system of the DSA-1 and to vary the key-controlled operating modes.

LEFT or L	'presses' the DSA-1 [<] key
RIGHT or R	'presses' the DSA-1 [>] key
ENTER or NEXT	'presses' the DSA-1 [ENTER/NEXT] key
EXIT	'presses' the DSA-1 [EXIT] key
EDIT or CLEAR	'presses' the DSA-1 [EDIT/CLEAR] key
SPKON	'presses' the DSA-1 [L.S. ON] key
SPKMODE	'presses' the DSA-1 [L.S.MODE] key
VOLUP or +	'presses' the DSA-1 [^] key
VOLDOWN or -	'presses' the DSA-1 [v] key
DIGI or DI	'presses' the DSA-1 [DI] key
TERM	'presses' the DSA-1 [TERM] key
CHAN	'presses' the DSA-1 [CHAN] key
DIGO or DO	'presses' the DSA-1 [DO] key
SYNC or SYNC_OP	'presses' the DSA-1 [SYNC O/P] key
HOME	performs enough EXITS to get to the top level of the menu system

7.4.2.3.2. Stores - reading the DSA-1 display

The Macro may need to obtain data from the DSA-1's display, for example to display or file a reading, or to range-check it. The Macro Processor has three stores for this purpose which are loaded using the following Keywords; the stores can be accessed by other Keywords for checking or displaying the stored data. The user can specify exactly where on the DSA-1's screen the data is to be read from, using a coordinate system: the top row is numbered 1, the bottom row 2, and the columns from 1 to 16 (left to right).

GET_DSA_TEXT(n, start, finish)	Read a text string from the nth row of the display, between the positions 'start' and 'finish' (inclusive).
GET_DSA_VAL(n, start, finish)	Read a numerical result using the same coordinate system as above. The number may be an integer or have a decimal point.
GET_DSA_CHAR(n, pos)	Read a single character from the nth row at position 'pos' using the same coordinate system as above. Note that not all DSA-1 characters translate exactly into PC characters - the bit activity display for example.

7.4.2.3.3. Branching

These keywords allow unconditional and conditional jumps to be made to another part of the macro sequence, depending on the state of the DSA-1 LEDs or the values in the internal stores (see above). They all specify a label to jump to:

Unconditional:

JUMP (@xxx) or JMP (@xxx) Unconditional jump;

On DSA-1 LEDs:

JUMP_IF_LED(zzz, @xxx) or
JLED(zzz, @xxx) Jump if the specified LED is lit;

JUMP_IF_NOT_LED(zzz, @xxx) or
JNLED(zzz, @xxx) Jump if the specified LED is unlit;

The following LED specifiers should be used with the above Keywords:

CARRIER panel LEDs: FAIL, ERROR, NFAIL, BLOCK, ALOG, 32, 44056, 441, 48, OTHER, 200PPM, 50PPM, 10PPM, 2PPM, WRTREF;

DATA panel LEDs: AUX, AUDIOLO, AUDIOHI, USER, VALID PRO, CONS, 1550, J17, ABDIF, CRCERROR, COPYPRO, STEREO, 2CH, MONO;

Key mode LEDs: ON, LS_ON, LSM_CHAN, LSM_BOTH, LSM_LINE, DI_XLR, DI_COAX, DI_OPT, TERM_LOZ, TERM_HIZ,

CHAN_A, CHAN_B, DO_LOOP, DO_RECLK, DO_FUNC,
SYNC_CH, SYNC_BLK, SYNC_REF;

Thus to jump if the sampling frequency was shown as 44.1kHz to a label '@good_fs', the Macro Command would be:

JUMP_IF_LED(441, @good_fs)

On contents of result stores:

JUMP_IF_CHAR('c', @xxx) or
JCHAR('c', @xxx)

Jump if the character specified is in the character store; the character must be enclosed in single quotes;

JUMP_IF_NOT_CHAR('c', @xxx) or
JNCHAR('c' @xxx)

Opposite condition to the above;

JUMP_IF_LESS(n, @xxx) or
JL(n,@xxx)

Jump if the value in the numeric store is less than n;

JUMP_IF_GREATER(n, @xxx) or
JG(n,@xxx)

Jump if the value in the numeric store is greater than n;

JUMP_IF_LESS_OR_EQUAL(n, @xxx) or
JLE(n, @xxx)

Jump if the value in the numeric store is less than or equal to n;

JUMP_IF_GREATER_OR_EQUAL(n, @xxx) or
JGE(n, @xxx)

Jump if the value in the numeric store is greater than or equal to n;

JUMP_IF_EQUAL(n, @xxx) or
JEQ(n,@xxx) or
JE(n,@xxx)

Jump if the value in the numeric store is equal to n;

JUMP_IF_NOT_EQUAL(n, @xxx) or
JNE(n,@xxx)

Jump if the value in the numeric store is not equal to n;

In all the above 'n' can be an integer or a floating-point number.

JUMP_IF_PASS(@xxx)

Jump if the global fail flag is clear, e.g. after a SET_PASS instruction;

JUMP_IF_FAIL(@xxx)

Jump if the global fail flag is set, e.g. after a SET_FAIL instruction;

7.4.2.3.4. Stores - reading the host PC keyboard

The Macro Processor's internal stores can also be loaded from the host PC keyboard, in order to allow conditional action on operator input. Ten text variables and ten numeric variables are also available.

GET_PC_TEXT(n)

Read text from the PC keyboard, delimited by **<Enter>**, and store in one of ten text variables specified by 'n'; n = 0 to 9. NOTE: These stores are additional to the DSA-1 text store loaded by GET_DSA_TEXT;

GET_PC_VAL

Read a numerical input from the PC keyboard, delimited by **<Enter>**. This is then held in the internal numeric store and may be checked in a conditional JUMP instruction;

GET_PC_CHAR

Read a single character from the PC keyboard into the internal character store. It may be checked in a JCHAR or JNCHAR instruction. NOTE: No **<Enter>** is required.

SET_VAR(n, val)

Copy the constant 'val' into numeric variable (n); n = 0 to 9;

STORE_VAR(n)

Copy the internal numeric store to numeric variable (n); n = 0 to 9. This Keyword is used to temporarily hold a value from the DSA-1 display or from the PC keyboard;

GET_VAR(n)

Copy numeric variable (n) to the internal numeric store; n = 0 to 9.

7.4.2.3.5. Host PC output

The following Keywords control output of data to the host PC's screen, LPT and COM ports and to disk files:

CLRSCR or CLS

Clear the PC screen;

PRINT(" ")

Display text on the PC screen;

The text between double quotes, up to 100 characters in length, is displayed on the PC screen, with the exception of tokens starting with a backslash ('\'). These have special actions:

\N	causes a new line;
\R	causes the cursor to move to the start of a line;
\T	causes a TAB;
\F	causes a FORM FEED - usually used with LPT_PRINT (see below);
\nnn	causes the ASCII character number nnn to be printed - note that this must have three digits, e.g. for an 'A' character use \065, and is specified in decimal notation;
\S	displays the text store (from GET_DSA_TEXT);
\V	displays the numeric value store (from GET_DSA_VALUE or GET_PC_VALUE);
\C	displays the character store (from GET_DSA_CHAR or GET_PC_CHAR);
\In	displays user-input text variable n (from GET_PC_TEXT(n));
\\	displays a backward slash character;
\"	displays a double quote character;

COM_OPEN

Open the auxiliary COM port (other than that used to communicate with the DSA-1);

COM_PRINT(" . . . ")

Print to the COM port currently open, tokens can be used as in the PRINT Keyword;

COM_CLOSE

Close the currently open COM port;

LPT_OPEN(n)

Open LPTn parallel port;

LPT_PRINT(" . . . ")

Print to the LPT port currently open, tokens can be used as in PRINT;

LPT_CLOSE

Close the currently open LPT port;

FILE_OPEN(" . . . ")

Open the filename in double quotes - the filename may include \I tokens (see above) so that, for example, entered serial numbers may be linked to filenames;

<code>FILE_APPEND("...")</code>	Open the file as above but, if the filename already exists, data is appended to the end of the file;
<code>FILE_PRINT("...")</code>	Print to the file currently open, tokens can be used as in <code>PRINT</code> ;
<code>FILE_CLOSE</code>	Close the currently open file;

7.4.2.3.6. Special actions

<code>SET_PASS</code>	Clear the global Fail Flag;
<code>SET_FAIL</code>	Set the global Fail Flag;
<code>SET_SPEED_FAST</code>	Sets the Macro Processor execution speed to the fastest possible (this is the default state);
<code>SET_SPEED_SLOW</code>	Sets the Macro Processor execution speed to a slower rate (this makes the operation of the DSA-1 easier to observe);
<code>BEEP</code>	Cause a short 'beep' from the PC;
<code>WAIT(n)</code>	Wait for n seconds before processing the next keyword (n need not be an integer);
<code>WAIT_ANY_KEY</code>	Wait forever for any PC key to be pressed;

WAIT_KEY(n, 'c')

Wait for n seconds or until the PC key specified by 'c' is pressed. If n is set to -1, then the Macro waits forever for the key.

In WAIT_KEY, instead of the 'c' parameter, these special tokens can be used (without quotes):

ENTER	the <Enter> key;
ESC	the <Esc> key;
SPACE	the space bar;

QUIT

End the macro immediately at this point;

7.4.2.4. Macro File programming tips

An important consideration for the reliable operation of Macro Files is ensuring that the DSA-1 will operate in the same way each time the Macro File is executed. It is therefore advised that the following steps are included at the start of any Macro File:

- i) Enter sub-menu 9.3. to select the DSA-1 'default' operating settings, and then alter any that you need to afterwards. This prevents the state of the DSA-1 on entry from affecting the operation of the Macro. Note that the start-up settings of the DSA-1 are not lost;
- ii) Use sub-menu 9.1. setting 5 to disable the DSA-1's sub-menu memory. This will allow Macro segments to be independent of each other since navigation of the menu system will not depend on what has happened before;

When navigating the DSA-1 menu system, specific sub-menus can be searched for by using a combination of GET_DSA_CHAR and JUMP_IF_CHAR statements to look for the menu numbers on the left of the DSA-1 screen. The same applies to parameter settings. However, if i) and ii) above have been observed at the start of the Macro File, the necessity for this is greatly reduced or removed.

Sometimes the DSA-1 may take a short time to perform a certain action, and during this time tokens sent to it by the Macro Processor may be ignored (for example, when selecting 'defaults' as recommended above). Therefore it is sometimes necessary to insert delays in the Macro File using the WAIT Keyword, to let the DSA-1 finish its operation before more tokens are sent.

Use comments liberally to make the file easier to read - especially if some of the shorter forms of keywords have been used.

The global Fail Flag can be used to indicate whether a particular test (or all tests) has passed or failed. This can be set or cleared during the macro sequence (initially it is clear) and tested using the JUMP_IF_FAIL or JUMP_IF_PASS Keywords.

7.4.2.5. Invoking the Macro Processor

The name of the Macro Processor program is DSAMAC.EXE. The program is invoked from the MS-DOS prompt by typing:

DSAMAC <filename> <options>

where <filename> is the name of the Macro File to be executed (including the extension), and <options> are one or more of the following:

- | | |
|----------|---|
| /1 or /2 | selects the COM port number used to communicate with the DSA-1. Note that the port number not used will be used as the 'auxiliary' port for all COM statements (see above); |
| /R | selects Macro Recording mode (see below); |
| /Bxxxx | sets the baud rate 'xxxx' for the auxiliary COM port; |
| /Cyxxx | sets the inter-character delay 'yyy' in milliseconds for auxiliary COM port output; |
| /S | selects Single Step mode for debugging (see below); |

During execution, errors are reported together with their line number, as the Macro Processor comes across them in the Macro File.

The 'return code' of the program (which may be checked in batch files using the 'if errorlevel' statement or by any other program that runs DSAMAC in a 'DOS shell') depends on the global fail flag and whether the macro sequence executed successfully:

0 is returned if the Macro File was executed without errors and the global Fail Flag was clear on exit;

1 is returned if errors were detected in the Macro File;

2 is returned if no Macro File errors occurred, but the global Fail Flag was set on exit;

The PC <Esc> key can be used at any time to stop Macro File execution.

7.4.2.5.1. Single Step mode

Single Step mode is invoked by using the '/S' switch in the DSAMAC command line. This mode is useful for debugging Macro Files.

In Single Step mode, one Keyword of the Macro File is executed each time a PC key is pressed. The programmer can thus step through the Macro File line-by-line to find problems.

7.4.2.5.2. Macro Recording mode

Macro Recording mode is invoked by using the '/R' switch in the DSAMAC command line. It provides a quick method of generating a 'skeleton' Macro File by operating the DSA-1 manually.

In Macro Recording mode, each time a key is pressed on the DSA-1 is translated into the appropriate Keyword and recorded to the specified filename. Text from the PC keyboard is also inserted into the file, so comments can be written marking DSA-1 navigation points in the Macro File. When this task has been completed, the resulting Macro File can be edited in the normal way to add complete the tests required.

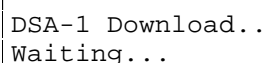
This feature can provide a speedy way of constructing a Macro File from scratch.

7.5. Downloading firmware updates

The DSA-1's operating firmware is stored in a 'FLASH' memory within the unit. This allows the code to be updated without the need to open the DSA-1 case, or to replace any ICs. Instead, the host PC downloads the firmware from a floppy disk, so updates can be easily circulated. The DSA-1 firmware downloader program is an MS-DOS application.

NOTE: The DSA-1 is supplied with the most up-to-date version of the operating firmware. Do not download the DSA-1 firmware unless you have received an updated version of the firmware!!!

Before running the loader, make sure that a connection is made between the DSA-1 and a COM port on the host PC using a null-modem cable (see section 3.6), and that the DSA-1 is powered from its DC adapter. Now turn on the DSA-1 **whilst holding down the [EDIT/CLEAR] and [DI] keys**. The DSA-1 should power-up with the message:



```
DSA-1 Download..  
Waiting...
```

If this message does not appear, power down and try again. Note that if only the upper line of the message is displayed, DC power has not been detected and the process must be restarted with DC applied.

When the message is displayed correctly, choose the directory in which the DOS support software is installed on the host PC (usually C:\DSADOS), and at the prompt type:

```
C:\DSADOS> dsaload dsav2p00.dld<Enter>
```

if the COM1: port is in use or, alternatively, if you are using the COM2: port:

```
C:\DSA1> dsaload dsav2p00.dld -2<Enter>
```

[NOTE: this example shows the filename of the version 2.0 download file; this filename will be different for subsequent versions.]

The DSA-1 display should then display a sequence of messages describing the erasing and block-loading of the FLASH memory. Finally, when loading is complete, the DSA-1 will start to run the new firmware. The PC should also confirm that the download has completed. If the download fails, make a note of any error message and contact your distributor. Note that even if the download fails, you can repeat the process since the 'boot' firmware in the DSA-1 is stored in an area that cannot be erased. If the DSA-1 detects that its firmware is incorrect, it enters the download mode automatically on power up.

It is wise to ensure that you have a copy of the existing firmware before updating, since you can reload the old firmware if you need to. Also, note that all setup options, User Test Sequences and the Results Log are automatically cleared as the new firmware starts for the first time.

8. DSA-1 test and measurement techniques

This section describes some typical ways in which the DSA-1 might be used. The discussions in this section are intended to provide a general understanding of the test and trouble-shooting strategies which are appropriate to different problems and situations, rather than an exhaustive list of the capabilities of the DSA-1 which are covered elsewhere in this manual.

A basic knowledge of the AES3 interface standard has been assumed, including the concepts behind various 'problem issues'. This subject is covered to an adequate level in appendix A, and the reader is referred here in case of difficulty. Likewise a familiarity with the operation of the DSA-1 is assumed.

8.1. System trouble-shooting

The later sections of this chapter are divided between equipment inputs, outputs, cabling etc. If you are in a system trouble-shooting situation, it is important first to establish where in the program chain the problem exists and whether or not an AES3 interface is to blame. It is equally often the case that some other equipment fault or operational error is responsible.

Even when a particular interface has been implicated, it may not be easy to know whether a carrier or a data problem exists, or whether the sourcing or receiving equipment or the cabling is to blame.

In general, intermittent and audio-degradation failures are generally carrier-related whereas 'hard' or total failures are likely to involve incompatibility of ancillary data such as Channel Status. However, there are many interesting exceptions which prove the validity of this rule!

If you suspect a particular AES3 interface, a good first step is to patch the DSA-1 'in-line' (connect the source to the DSA-1's appropriate DI, and the DSA-1's DO to the receiver). If you have significant cabling, patch-fields, Krones etc. in the path, it is best to patch the DSA-1 in just before the receiver.

Having patched in the DSA-1, select the appropriate input and see if the DSA-1 can receive the signal. The DSA-1 can receive carriers which would be unacceptable to most other equipment, and it can receive data irrespective of ancillary data such as Channel Status and Validity flags. If the DSA-1 can't receive the carrier (i.e. the **'FAIL'** LED remains lit) then a catastrophic source carrier fault is indicated; however, before assuming this, repatch the DSA-1 at the transmitter side of any cabling, patch-fields, Krones etc. to eliminate these first. If it still looks like a source equipment problem, refer to section 8.2 below. Otherwise, if the source now appears sound without all the interconnections in between, try section 8.5.

Alternatively, if the DSA-1 can lock on the carrier successfully when patched in before the receiver, you can monitor using headphones or the built-in loudspeaker to check that the audio signal is intact. If the audio can be monitored correctly by the DSA-1 but not by the receiving equipment, try switching the DSA-1 into **'RPT'** mode with the **[DO]** key. This regenerates a de-jittered, full amplitude carrier at the DSA-1 output.

If **'RPT'** mode allows the receiver to work properly, a source carrier problem or a cabling problem is indicated; refer to the appropriate section below. If **'RPT'** mode doesn't help and the problem seems to be carrier-related, then a receiver fault is indicated since the DSA-1 output in **'RPT'** mode is of good quality. However, it remains possible that some sort of status problem is to blame, particularly if the receiving device is exhibiting a 'hard' fault such as no lock or no audio. Obvious Channel Status inconsistencies or the 'invalid' state of the Valid bit should be checked for on the DSA-1's DATA panel.

If none is found, the next step is to switch the **[DO]** key to **'FNC'** (function generator) mode, set the generator Sync Source to DI, and select GEN1. This mode allows the DSA-1 to regenerate a de-jittered output as before, but with complete control of all Channel Status fields and the Valid bit. Details of how to use this mode to test a receiver are covered in section 8.3 below. If the receiver can be made to work successfully by forcing Valid, then a source-invalid fault is indicated; if editing the Channel Status does the trick, then a Channel Status compatibility fault must be the cause. It may be difficult to sort out which end is to blame for this, although tracking down the relevant field usually makes it clear. Alternatively, running the appropriate automatic Test Sequence as described in the next section should verify the Channel Status compliance of the source.

The most difficult system faults to diagnose are those which are intermittent. The DSA-1 Watchdog mode is designed to help track these down. This is described in the following section. Note that the Watchdog mode can be set up with the DSA-1 'in-line', so that intermittent events can be logged over long periods whilst the system is operating normally.

8.2. Evaluating equipment outputs

The evaluation of the compliance of AES3 sources was the main function of the DSA-1 in its early form. In those days it was common to find sources which, for one reason or another, were not acceptable to inputs of other equipment. It was perceived at the time that non-compliance of sources was a much greater problem than non-compliance of receivers. This is probably still as true as it was, although now the capabilities of the DSA-1 encompass receiver testing as well.

The most frequent reasons for source-related problems are poor carrier generation and Channel Status incompatibility. Cabling problems often confuse the issue, but are not considered as source problems here, and are dealt with in a later section. Audio-data issues are not generally considered to be the domain of the DSA-1, for these we

suggest the use of the Prism Sound Dscope system; however, the DSA-1 does deal with some audio data issues, for example word-length compatibility.

8.2.1. General source trouble-shooting

Having connected the source to the appropriate DSA-1 DI input, the first thing to check is whether the DSA-1 can lock to the source. This is indicated by the **'FAIL'** LED going out and the CARRIER and DATA panels becoming active.

If the DSA-1 cannot lock, a serious carrier problem is indicated since the DSA-1 can generally lock on carriers which most equipment would not accept. First of all eliminate any cabling by connecting the source straight to the DSA-1 with a different cable. Some DSA-1 analysis functions are active even if the input is unlocked in order to help in this situation. Check the carrier amplitude - if this is zero or very low, then the output may be damaged or switched off (some equipment allows the digital outputs to be disabled). If you are using balanced AES3, test the common-mode amplitude. If this is more than a few hundred millivolts then the source signal probably has an open-circuit or short-circuit leg. In the case of a transformer-coupled output, the latter is indicated since an open-circuit would result in loss of the differential amplitude reading. If these readings seem to be reasonable, some sort of catastrophic content-error is implied, for example incorrect AES3 preamble/biphase coding or perhaps operation at a wildly incorrect sampling rate. These can be investigated by connecting an oscilloscope to the Test Out connectors of the DSA-1.

If lock is successfully achieved, a good way to start is to run the appropriate automatic Test Sequence. In the case of professional equipment, Sequence Sys04, the 'Strict Professional Test' is recommended. This test requires a source compliance which is rather more strict than the AES3 standard (which does not generally insist on operating margin between transmitter and receiver) but which should be achievable by most professional devices. Most aspects of carrier and status compliance are quickly tested, and any problems are highlighted.

To detect intermittent source problems, the DSA-1 Watchdog mode should be used. This allows long-term logging of any CARRIER and DATA panel activity. For example, intermittent biphase errors, block-length or carrier phase errors, sample rate changes or drifts, or changes in Channel Status or Valid flags can all be captured and time-tagged.

8.2.2. Source carrier tests

The **sampling-rate** accuracy of a source should be checked if the source is operating as a clock master. This is usually only relevant in determining whether other equipment will have sufficient range to lock to the source. Contrary to popular belief, there is no connection between the absolute frequency of a synchronisation source and its 'stability' or jitter performance, which may affect the audio quality of non-jitter-attenuating conversion equipment synchronised to it. In general, good quality professional equipment should provide master sync within $\pm 10\text{ppm}$, and lower grade professional equipment within $\pm 50\text{ppm}$. Equipment designed to provide a reference clock should generally be within $\pm 1\text{ppm}$, and thus will usually be more accurate than the DSA-1's own reference, unless the DSA-1 is fitted with the $\pm 0.5\text{ppm}$ reference option or has recently 'learned' an accurate reference source. Low grade consumer equipment is generally permitted to be up to $\pm 1000\text{ppm}$ adrift when clocked internally, although it is very rare nowadays to find examples anywhere near this limit.

The **carrier amplitude** of a source should be checked to be within reasonable limits according to its electrical output format. However, it should be noted that a very wide range of voltages is permissible - in practice, lack of carrier amplitude is almost never the cause of a problem, unless long and lossy cables are used, in which case data jitter would probably dominate in any case.

Likewise the **source impedance** can be checked, but is unlikely to be responsible for any practical failures. Early equipment was often not source-terminated at all, and generally presented no problems. Although mismatched impedances can cause significant visible distortion of the carrier waveform, this is irrelevant unless it affects the transition timing of the sliced data at the receiver. For shorter cable lengths this is unlikely since the rise and fall times are fast and the peak-voltages are well above the slicing threshold. Where long, lossy cables are used, reflection artifacts are not generally observed since they are effectively absorbed by the journey back up and down the cable. Having said this, one occasionally encounters horrific installations where particularly unfortunate short cable lengths exactly match some critical wavelength, or else a cable type of such low characteristic impedance (for example 'starquad') has been used, that problems have resulted.

By measuring the amplitude of the carrier with the DSA-1 set first to LO-Z and then to HI-Z, a source impedance factor can be approximated:

$$\text{Impedance factor} = (V_{\text{HI-Z}}/V_{\text{LO-Z}}) - 1$$

The actual source impedance is the impedance factor multiplied by the correct impedance for the selected DI format. For example, if $V_{\text{HI-Z}}$ reads 4.5V and $V_{\text{LO-Z}}$ reads 3.1V for the XLR DI input, the impedance factor is $(4.5/3.1)-1$, which is about 0.45. The source impedance is therefore $0.45 \times 110\Omega$, or 49.5Ω . Obviously for a correct source impedance, $V_{\text{HI-Z}}$ would be twice $V_{\text{LO-Z}}$.

Intrinsic jitter from a source is measured using the 'fs Jitter' sub-menu. There is no proper specification for this parameter in the standard (see the footnote at the end of this chapter). 40ns P-P is suggested (about 1/4 UI at fs=48kHz) but any source supplying intrinsic jitter at this level would probably be beyond the grasp of some receivers. It is unusual to see intrinsic jitter figures above 20ns these days, and most good equipment will manage under 5ns. Any moderate figure should be acceptable, except where non-jitter-attenuating converters are to be connected, in which case the figure should be as low as possible, although the problem actually lies with the converter. Remember to also check the intrinsic jitter with the source externally synchronised to a low-jitter reference, if this is a system option. This can be done by using the DSA-1's GEN2 (in internal Sync Mode) as the reference. On poor quality equipment, the intrinsic jitter can be very much worse on external sync.

The **eye-narrowing** and **data jitter** readings should also be checked, mainly to verify that none is present. These are cable effects and a source should not register anything here if the cable to the DSA-1 is short, unless some distortion or band-limiting is present in the AES3 output driver. Eye-narrowing should be zero, and data jitter should be nominally equal to the previous fs jitter result. Likewise the **biphase error count** should continue to register zero with a short cable unless the equipment is faulty.

If the source can be synchronised externally, the **phase** of its carrier output should be checked with respect to the reference input using the 'Phase v. Ref' sub-menu. This figure should be less than $\pm 5\%$, although it is unlikely that any modern receiving equipment would suffer audible problems whatever the output phase of the source, since reframing hysteresis is usually included in AES3 receiver devices. Check the figure for Wordclock as well as AES11 reference syncs, if these can both be used.

8.2.3. Source data tests

The correct operation of the **Channel Status** and **Valid** bits from the source should be checked, although it is often hard to know exactly what to expect. Simple equipment should issue 'Valid' (Valid bit = 0, DSA-1 '**V**' LED OFF) all the time, although non-audio or compressed audio sources should show 'Invalid'. Uncorrectable errors or the STOP or PAUSE state in some transports may also result in 'Invalid' indications. Various levels of Channel Status implementation are now allowed although, in general, professional equipment should at least have all fields in the first three bytes set correctly plus a correct CRC to assure trouble-free interfacing. Consumer equipment should correctly implement all fields in the first byte, and indicate the correct sampling rate in byte 3. A non-copyright category code and 'copy permitted' should also be set if the source is to be recorded on SCMS devices.

The **audio word-length** of the source can be checked using the DSA-1's 'active bits' sub-menu. If the source equipment contains any digital processing, the output re-dithering or noise-shaping can be examined using the DSA-1 Microscope mode as described in section 8.4 below.

Finally, the DSA-1's audio meter (in numerical readout mode) can be used to check the **line-up level** of the source if a line-up tone is available in the system.

8.3. Evaluating equipment inputs

It is not uncommon to encounter AES3 inputs on some equipment which seem abnormally fussy about which sources they will accept. To establish whether the AES3 input is actually at fault, it is necessary to test its response to a variety of input carrier and data situations. If the input is habitually connected via particular cabling, patch-fields or Krones, these should be suspected first. It is quite common, where carriers are close to failure, for inputs to have difficulty with initial locking but to work adequately once lock has been achieved.

8.3.1. General receiver trouble-shooting

A good general test is to use the DSA-1 GEN2 to inject an fs/4 tone into the input under test. The GEN2 Channel Status output should be set to 'professional' or 'consumer' according to the requirement of the input. The DSA-1 should generally be set to Internal Sync, but it may need to be locked to the 'house-sync' if that is expected by the input under test.

If the input cannot receive the DSA-1 test tone, eliminate any cabling etc. by connecting the DSA-1 straight into the input with a different cable. If this does not cure the problem, try selecting the DSA-1's GEN1 instead, since this allows individual editing of the output Channel Status fields. A single unexpected field is often enough to cause an intolerant input to 'play dead'. Although the standard Channel Status settings of GEN2 should be acceptable to any professional or consumer input, it may be that the suspect input expects some unusual field setting(s), which you can set using GEN1. If the suspect input can lock to any other piece of equipment, it may be easier to 'capture' the Channel Status of that equipment using the 'C-S Capture' sub-menu and then to successively try 'standardising' the Channel Status fields until the input fails.

If the input can receive the GEN2 output successfully, the fs/4 tone is a useful signal to monitor for a while, since it is of high enough frequency to render any sample-repetition or omission (through sample-slipping) or any sample-repetition or interpolation (through received errors) easily audible.

8.3.2. Receiver carrier tests

The **jitter tolerance** of an AES3 receiver is the jitter amplitude which can be present on its input carrier before the receiver becomes incapable of decoding the data. The jitter tolerance is not a simple value since it always varies with jitter frequency - all receivers can tolerate large amounts of low-frequency jitter, but few can tolerate more than 1/2 UI (80ns P-P at fs=48kHz) at high frequencies, since the biphasic-mark coding

has become very indistinct by this point. The interface standard is unhelpful on the subject of jitter tolerance (see the footnote at the end of this chapter) but suggests 40ns P-P should be achieved at no particular frequency.

To fully characterise jitter tolerance, by applying continuously variable amounts of jitter at any desired frequency, a Prism Sound JM-1 Jitter Modulator is required. However, the DSA-1 can provide a useful set of jitter conditions at its output which allow jitter tolerance to be generally assessed. By selecting the DSA-1's GEN2 output, jitter can be applied in 1/8 UI steps from 0 to 3/8 UI (about 0 to 60ns P-P at $f_s=48\text{kHz}$) and with a variety of jitter functions (waveforms). In general, a rugged input will tolerate the full 3/8 UI amplitude with any function. Any input that will not tolerate 1/4 UI amplitude with all functions is notionally outside the rather vague specification currently available. Note that the $f_s/8$ and PRS-L (low-frequency noise) functions are less demanding than the cable-simulation and PRS-H (wide-band noise) settings.

The 'Cable' setting of the DSA-1 jitter generator allows receivers to be tested with a jitter function which emulates a long or poor-quality cable. In this mode, data jitter is added by emulating the intersymbol interference of a cable.

By connecting the DSA-1 'in-line' and selecting GEN2 and the DI sync mode, it is possible to add controlled jitter to a carrier in-system. This is useful for evaluating the operating margin.

8.3.3. Receiver data tests

The problem of an AES3 input failing through Channel Status incompatibility was mentioned above. Beyond basic functionality, it may be useful to check that equipment responds correctly to relevant **Channel Status input**. This is simply done with the DSA-1's GEN1 mode, which allows all Channel Status fields to be individually defined. It is not always clear what the desired response to a particular Channel Status setting should be, but it is generally adequate to check the correct operation of necessary actions and freedom from common false actions, according to the needs of the installation. Here are a few suggestions:

- | | |
|--------------------------|---|
| For professional inputs: | Does 'non-audio' cause muting? |
| | Does 'emphasis' type cause correct de-emphasis? |
| | Does 'unreliable' override the above? |
| | Does 'CRC error' override the above? |
| | Does 'CRC error' cause spurious muting? |
| For Consumer inputs: | Does 'non-audio' cause muting? |
| | Does 'emphasis' type cause correct de-emphasis? |
| | Does 'copy enable' or SCMS work? |

Correct operation of the **Valid bit** should be checked in the same way although, again, a variety of 'correct' responses are possible since the Valid bit may have different meaning for different source devices. In general, one would expect receiving equipment to either mute or sample-hold in response to an 'Invalid' flag. Some equipment implements a sample-hold which gradually decays to a mute - this is a clever idea since it allows concealment of single-sample errors as well as muting of prolonged 'Invalid' periods.

Finally, the DSA-1 can be used to inject precise **line-up** tones into the input under test, to check gain settings, level meters etc. GEN1 also features a 'Pulse' waveform which allows digital overload indicators to be tested with periodic full-scale pulses of one, two, three or four samples duration. The pulses can be positive-going, negative-going or both.

8.4. Evaluating 'in-line' performance

It is useful to test digital-to-digital equipment 'in-line' since this may provide more information than testing the inputs and outputs separately as described above.

8.4.1. 'In-line' carrier tests

It is especially important to perform the following tests if equipment is 'cascaded', i.e. if devices are connected 'in-line' and each is synchronised from its data input. In this situation the tests can be usefully applied to the whole chain at once, since carrier degradation in cascade is normally progressive. The following tests are made by connecting the DSA-1's appropriate GEN2 output to the input of the system under test, and the output of the system under test back to the appropriate DI input of the DSA-1.

The DSA-1 generator would normally be set to Internal Sync. By selecting the fs/4 GEN2 function and monitoring using headphones or the built-in loudspeaker it is easy to hear sample-slipping or interpolated errors in the equipment or cascade.

Jitter attenuation is the amount by which jitter entering a device is attenuated at the device output. Jitter attenuation is not a simple value since it always varies with jitter frequency. The interface standard does not cover the subject of jitter attenuation (see the footnote at the end of this chapter). Most equipment may be categorised as a jitter attenuator, or not. A jitter attenuator reduces incoming jitter even at very low frequencies (sometimes down to a few Hz). Equipment which is not specifically a jitter attenuator normally reduces jitter outside the audio band (say above 50kHz) since this is usually a characteristic of the AES3 receiver device itself. To characterise jitter attenuation precisely, by applying constant jitter amplitude at any desired frequency, a Prism Sound JM-1 Jitter Modulator is required. However, the DSA-1 can make a general assessment of jitter attenuation by applying jitter to the input under test and measuring the output fs jitter. The two useful jitter functions (waveforms) generated by

the DSA-1 are the PRS-H (wideband noise) function and the fs/8 function. The latter is significantly reduced only by equipment which attenuates jitter within the audio band.

If the **carrier phase-shift** through the equipment under test exceeds 25%, the **'BLOCK/PHASE'** LED on the DSA-1 is lit. To measure the carrier phase shift directly, it is necessary to lock the input to the test equipment to a separate reference which is also connected to the DSA-1 Ref Sync input. The phase can then be measured in the 'Phase-v-Ref' sub-menu, with the DSA-1 generator off.

The **data integrity** of a suspect equipment or cascade can be monitored using the DSA-1 'Channel Check' mode, if the data can be passed through unprocessed. Channel Check mode tests the correctness of every bit of every audio sample for as long as required. This mode can also be used to check digital audio recorders by recording the test pattern prior to running the Channel Check. Although this is strictly a data test, intermittent data errors are usually caused by a marginal carrier.

8.4.2. 'In-line' data tests

By using GEN1 instead of GEN2 in the above setup, it is possible to test aspects of data transparency.

Channel Status transparency is often an important system issue, since a lot of equipment, whilst generating its own Channel Status output, is not adequately transparent to Channel Status from its input. Thus Channel Status applied by a system source is often destroyed by a subsequent device. GEN1 allows the Channel Status fields to be defined individually, and the 'C-S Fields' sub-menu in 'Data Tests' allows the returned fields to be examined.

The **SCMS** operation of a consumer recorder can be tested by setting the various SCMS states in GEN1, recording the data, and monitoring the returned SCMS status on playback. The DSA-1 displays the following SCMS states in its 'C-S Fields' sub-menu:

SCMS:Orignl Matl	The category code indicates that the material is from an original source. DAT ID6 is set to x0.
SCMS:Copy, gen1+	The generation status of the material is 1 or higher, e.g. it is a home-copy of original material. (DAT ID6 will be set to x1).
SCMS:Copy, gen??	The category code of the material provides no 'generation status'; e.g. signals from some A-D converters.

Lack of **user data transparency** can cause similar problems to the Channel Status case. Although the DSA-1 cannot generate user data, it can be useful in tracking down the point of user data destruction by use of the '**U**' LED in the DATA panel.

The **audio gain** can be tested by setting a known output amplitude for GEN1, and checking the returned amplitude with the audio meter in dBFS readout mode. Since the meter records peak-sample amplitude, an amplitude should be chosen some way below clipping to avoid error. **Frequency response** can also be assessed by this technique by making spot measurements at different GEN1 frequencies, although this task is better tackled with a Prism Sound Dscope system.

If the equipment or cascade under test applies any digital signal processing, the output **re-dithering or noise-shaping** can be examined using the DSA-1 Microscope mode. To do this, a separate source should be applied to the equipment input with a low-level test signal near to the re-quantisation noise floor (or the equipment itself can be used to attenuate a nominal-level input if it is capable). By applying gain to the equipment output using the Microscope, the low-level signal can be monitored using headphones or the built-in loudspeaker (or by monitoring the DO output of the DSA-1). The best test signal is one with a steady decay, such as a synthetic 'bong' or solo piano music. If the re-dithering or noise-shaping is adequate, the decay sounds clean, with the signal disappearing into smooth and continuous noise. If the signal breaks up or buzzes as it decays, or if the noise is modulated by the decaying signal, inadequate dither level or an inappropriate dither noise source is indicated. The Microscope input mask should be set to the appropriate word-length in case extra data is not truncated at the output under test. If the dither noise is not audible beneath the test signal, then the test signal amplitude should be reduced. If the Microscope overload indicators are activated, the Microscope gain should be reduced.

8.5. Evaluating AES3 cabling and interconnects

The carrier-degradation of any interconnection, including cabling, patch-fields, Krones etc. can be tested with the DSA-1, either with the normal system AES3 source in place, or by stimulating the input from the DSA-1's GEN2. The two significant measurements are 'Data Jitter' and 'Eye Narrowing'.

The **Data Jitter** measurement assesses jitter in sections of the carrier which are subject to data-dependent variation. The measurement represents the total jitter, including any fs jitter present from the source (although this is normally minimal in comparison to the cable-induced jitter over moderate to long ranges). Since the parameter is data-induced, it is important to remember that the program data affects the reading, so a representative audio signal should be used. If GEN2 is the source, the PRS (white noise) setting is the best. Although the interface standard is vague on all jitter-related matters (see the footnote at the end of this chapter) a figure of 40ns P-P is implied for worst allowable jitter. In the case of data jitter, which is normally predominated by high frequencies, this is a reasonable limit to apply since most receivers will manage a certain amount more.

The **Eye Closure** measurement is a variation on the same principle. It measures the narrowing of the eye of the narrow pulse in the X-preamble, rather than in the audio data areas of the carrier. This is preferable for two reasons: The narrowing of that particular eye is normally the worst of any eye in the carrier, since it is a 1UI pulse immediately following a 3UI pulse and the intersymbol interference effect is dominated by this ratio. Also, since this eye is not in the audio data area of the carrier it is largely unaffected by the audio data. So the Eye Closure measurement is a worst-case figure which can be made independent of program content. Like the Data Jitter reading, the Eye Narrowing reading is also inflated by fs jitter from the source. The 40ns implied limit from the interface standard is the same as for data jitter.

The **data integrity** of the AES3 carrier on the interconnect can be checked using the DSA-1 'Channel Check' mode, which tests the correctness of every bit of every audio sample for as long as required. This test can be done using the DSA-1 to drive and receive the interconnect, but this method is not very useful since it tests the ability of the DSA-1 to transact data over the interconnect rather than that of the actual equipment normally connected to it. It is therefore preferable to inject and extract the test pattern through the normal driver and receiver if possible; this is straightforward where reclocking routing switchers or distribution amplifiers are installed, but may be less easy if equipment is cabled directly.

For details of evaluating cabling for its effects on sampling jitter in conversion devices, see the following section.

8.6. Evaluating A/D and D/A converters

This section contains a brief summary of the issues of sampling jitter; these are covered in depth in the jitter papers listed in appendix D.

Assessment of the performance of A/D and D/A converters is a complex issue, and one for which the DSA-1 as a carrier generator/analyser is not intended. Audio signal generator/analysers such as the Prism Sound Dscope system are appropriate to this task. However, there are carrier-related performance criteria which apply to converters.

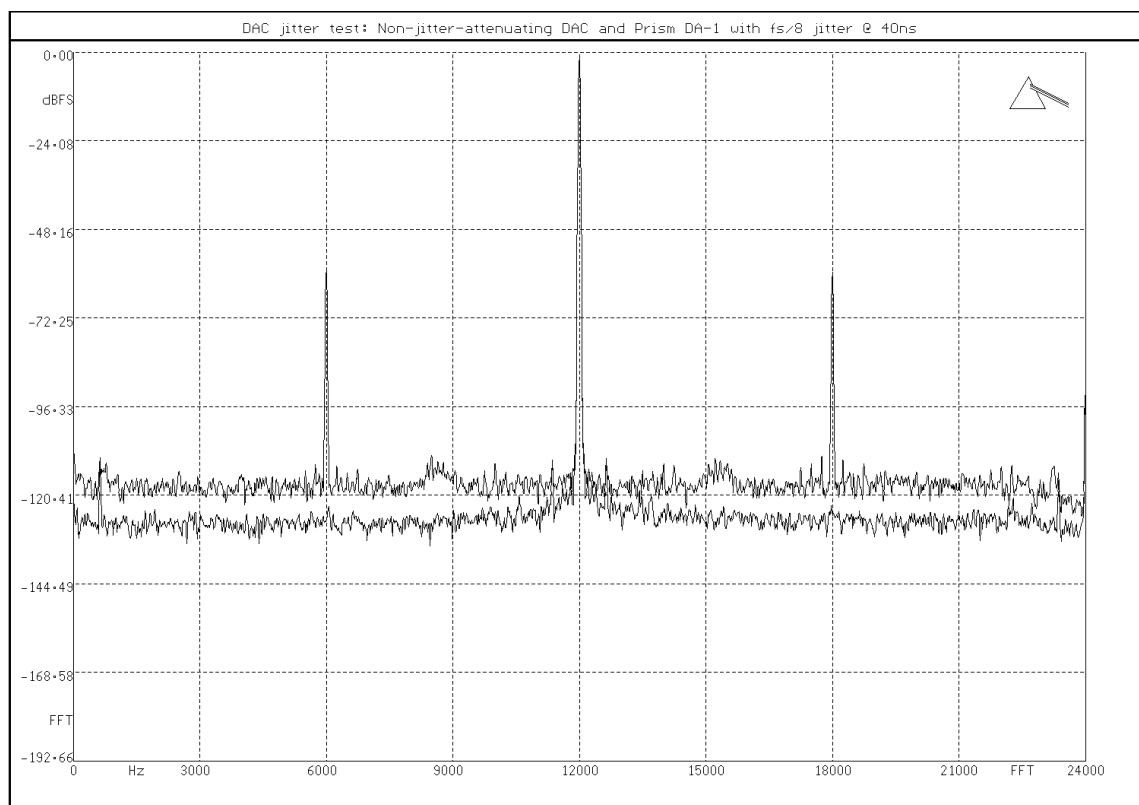
The main area is that of sampling jitter, whereby audio distortion is introduced at an A/D or D/A conversion owing to jitter in the clock of the converter sub-system itself. Since, in systems, A/D and especially D/A converters rarely generate their conversion clocks from a local oscillator, but rather from an AES3 or AES11 external reference, there is considerable scope for interface jitter to cause sampling jitter. However, it should be remembered that the two are not necessarily directly connected, and proper design of conversion equipment can effectively prevent the connection. But at present, most conversion equipment is not properly designed in this respect and for most, any audio-band jitter applied to the reference input will appear unattenuated at the converter sub-system.

This has resulted in different AES3 cable types and synchronisation schemes being associated with sound quality. The cable types and the interface standard itself are blamed for degradation rather than the inadequate converters. Clearly, within the converter, this is a problem of jitter attenuation, like that already discussed for in-line digital equipment. However, the design requirement is a much more stringent one since jitter amplitudes which can produce identifiable sampling distortion are orders of magnitude lower than those which would cause loss of data on the interface.

Some other common misconceptions have arisen from this confusion between interface jitter and sampling jitter; for example there have been moves within consumer circles for the jitter limit on the interface to be reduced to a level where it would not cause appreciable sampling distortion if unattenuated in converters. This is simply not possible for any appreciable transmission distance with a realisable cable and drive electronics - nor is it necessary. Others have taken to inferring the performance of conversion equipment which has both digital inputs and outputs by measuring the jitter on the digital output and assuming it to be the same as in the converter subsystem, which it probably is not. Perversely, it is probably impossible to measure jitter in the time-domain anywhere near as accurately as can be done by applying it to a conversion sub-system converting a known audio signal and measuring the distortion products.

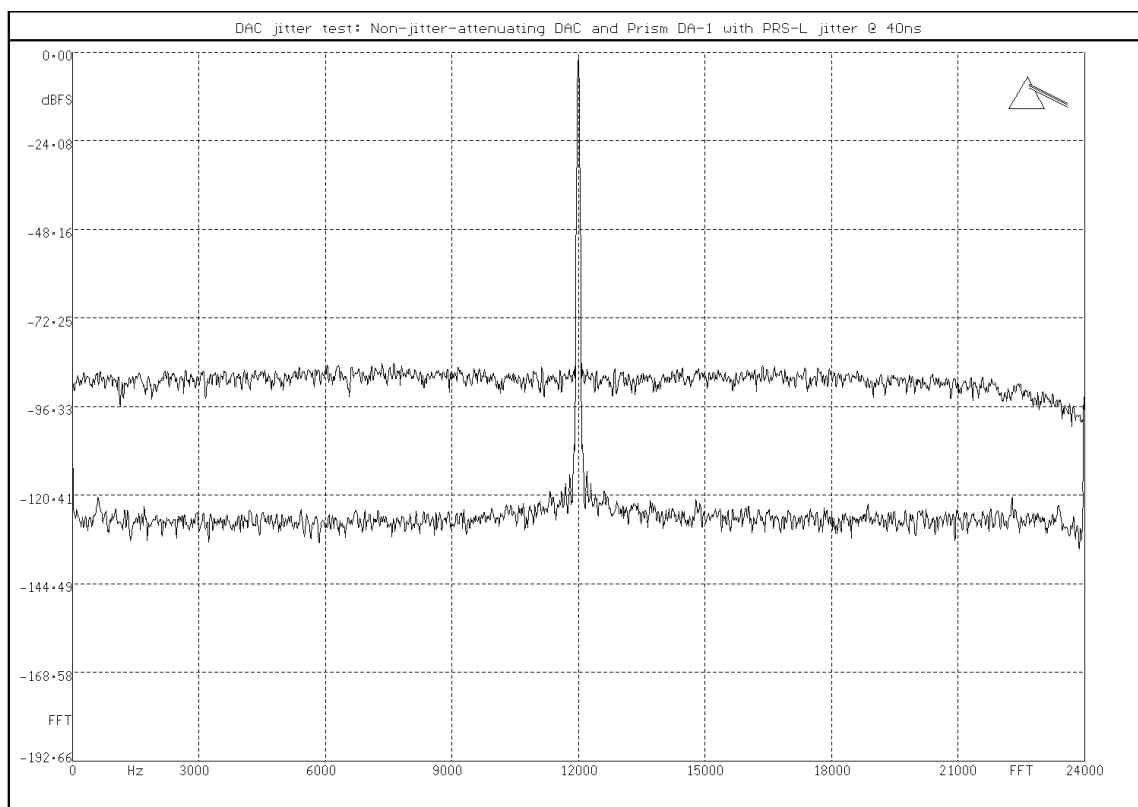
The DSA-1 can be useful in testing the susceptibility of converters to interface jitter, since it can simultaneously generate audio and jitter waveforms which are useful for this purpose. It cannot, however, assess resulting audio distortion and a conventional analogue or digital audio generator/analyser is necessary for this.

To test the jitter susceptibility of a D/A converter, the DSA-1's GEN2 is connected to the converter input with the fs/4 audio function selected. Jitter is then applied using the fs/8 jitter function. An FFT analysis of the converter output is shown below for a non-attenuating D/A converter and for a Prism Sound DA-1, which has extreme jitter attenuation, each with 40ns P-P jitter amplitude (the nominally allowed value in the interface standard):



The sum of the side-bands resulting from the sampling jitter for the non-attenuating D/A is only 56dB below the -1dBFS test signal. Although this unit is nominally an 18-bit converter, 40ns of interface jitter has produced distortion components which degrade the THD+n performance to that of a 10-bit system. The jitter frequency is $f_s/8$ (6kHz at $f_s=48\text{kHz}$) since this falls conveniently in the lower half of the audio band. This test can therefore be used to establish how much jitter attenuation is exhibited at $f_s/8$ by measuring the amplitude of the sum of the jitter side-bands and comparing it to the above example for zero-attenuation. The test can be done with an analogue distortion analyser if no FFT analyser is available.

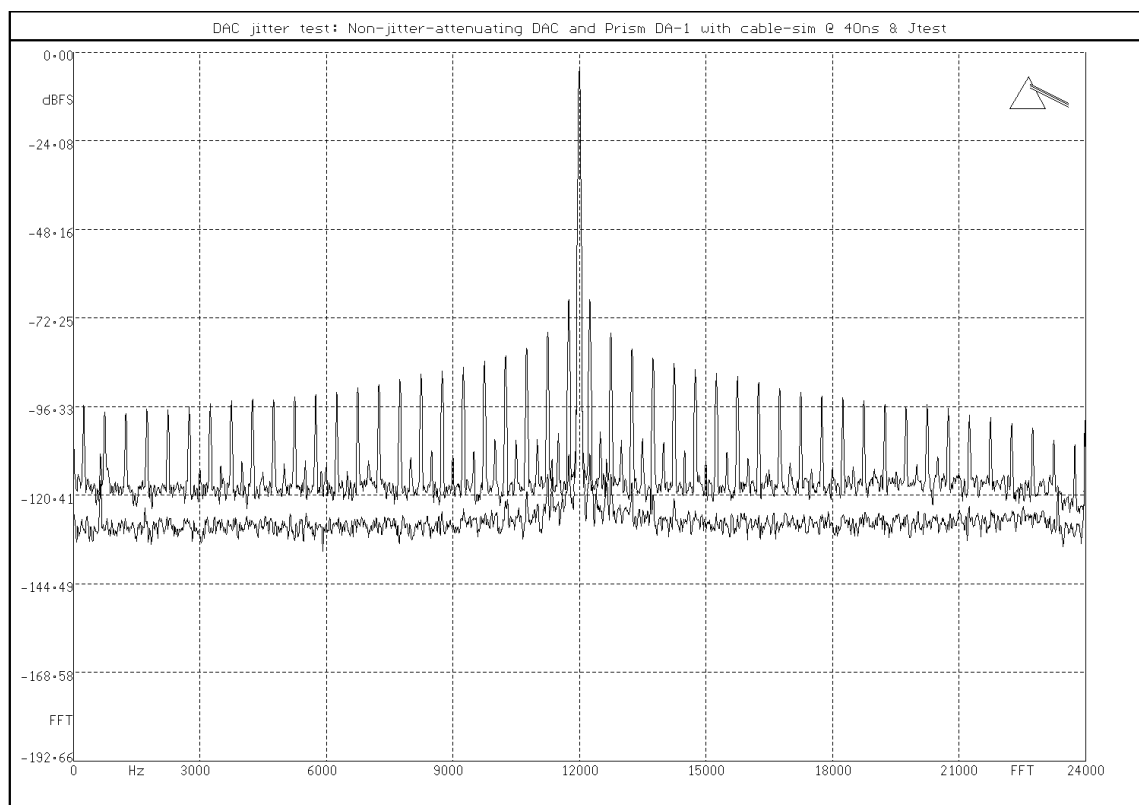
If the test is repeated using the DSA-1's PRS-L (audio-band noise) jitter function, the result is:



In this case, the signal-to-noise ratio of the non-attenuating converter has been reduced to 60dB by the application of 40ns P-P of audio-band jitter noise. This test also can be done with an analogue distortion analyzer, although the precise result may depend on the bandwidth of the analyser notch-filter. This test has the advantage that it can show the frequency response of the jitter attenuation of the D/A converter in the audio band. In this case, the trace is flat because there is no attenuation at any frequency.

The above tests can also be applied to A/D converters by applying the DSA-1 jitter generator to the reference input of the converter and an analogue $f_s/4$ tone just below clipping at the analogue input. If a digital FFT analyser is not available, the digital product could be returned to analogue for analysis providing a known highly-jitter-attenuating D/A converter is used.

The DSA-1 can generate a special audio waveform called Jtest, which was specifically developed by Prism Sound to excite extremes of data jitter on a cable. It contains principally an $f_s/4$ tone, but the DC-level of the tone is very slightly altered every 192 samples to produce a square jitter modulation at $f_s/192$ (250Hz at $f_s=48\text{kHz}$) according to the degree of intersymbol interference exhibited by the cable. The FFT result below shows the same two D/A converters, but this time driven with the Jtest waveform. 40ns P-P of DSA-1 cable-simulating jitter has been applied:



This test would normally be applied to a D/A converter though its normal cabling; the Jtest effectively is a test of both the cable and the D/A converter. THD+n for the non-attenuating D/A converter above is -73dB.

Finally, it should be noted that the AES3 output and input tests detailed at the start of this chapter apply equally to the AES3 outputs and inputs of A/D and D/A converters respectively.

8.7. Analogue audio pathfinding

The DSA-1 can also be used to verify analogue audio paths.

Monitoring of analogue signals is possible using headphones or the built-in loudspeaker, by setting the **[LS MODE]** to **'LINE'**.

No dedicated analogue generator is provided, but the digital generator can inject a passable analogue tone, even into a low-impedance line, by selecting the Jtest function (which has a strong 250Hz component). For higher impedance lines it is possible to use the GEN1 Squarewave function set to an amplitude of -20dBFS. This allows the frequency to be varied in the usual way.

NOTE: Concerning jitter parameters in the interface standard:

There are currently no proper specifications in the interface standards for any of the jitter parameters discussed above (intrinsic jitter, jitter tolerance, jitter attenuation), except to say that all transitions on the interface should always appear within $\pm 20\text{ns}$ (40ns P-P) of their 'correct' time. Conspicuously absent is a safety margin between the maximum jitter permissible from a source and the jitter tolerance of a receiver, as well as any definition of the LP filter characteristic to be used in determining the 'correct' time of the transition. An extension to the AES3 standard specifying jitter-related parameters is currently under consideration - this is listed in appendix D.

9. Specification

The following specification applies to DSA-1 units with hardware revision C or above, firmware revision 2.0 or above. Unless otherwise stated, quoted performances refer to XLR carriers at $f_s=48\text{kHz}$. The specification is offered in good faith but is subject to alteration without notice.

HARDWARE

I/O connectors:

DI (digital audio input) formats:	110 Ω balanced (AES3, IEC958, 'AES/EBU') 75 Ω coaxial (IEC958, 'SPDIF', or AES3-ID) Optical TOSLINK (CP340)
DO (digital audio output) formats: (loop-thru, reclock and generator)	110 Ω balanced (AES3, IEC958, 'AES/EBU') 75 Ω coaxial (IEC958, 'SPDIF', or AES3-ID)
REF SYNC input formats:	110 Ω balanced (AES3/IEC958/AES11 or 'Wordclock') 75 Ω coaxial (AES3/IEC958/AES3-ID or 'Wordclock')
Balanced connectors:	3-pin 'XLR', inputs female, outputs male
Coaxial connectors:	BNC, with phono (RCA) adapters provided
Input & output impedances:	$\pm 4\%$
Regenerated DO amplitude:	XLR: 4.4V P-P $\pm 5\%$ into 110 Ω Coaxial SPDIF: 0.5V P-P $\pm 5\%$ into 75 Ω Coaxial AES3ID: 1.0V P-P $\pm 5\%$ into 75 Ω
DI/REF SYNC f_s range:	28kHz - 52kHz
DO internally-generated f_s :	44.1kHz, 48.0kHz to TCXR accuracy
Oscilloscope (Test Out) outputs:	LINE and SYNC: x10, 75 Ω , BNC, transformer-isolated
RS-232 serial port:	PC-AT-type 9-pin male 'D' connector, 1200-9600 baud, no parity, 8 data bits, 1 stop bit, galvanically isolated from measurement circuits
DC adapter input:	9V to 12V @ 1000mA, 2.1mm DC 'barrel' socket, positive on inner

Analogue

Monitoring D/A converter:

THD&n : 0.01%, -80dB (@1kHz,-0.5dBFS)
 Frequency Response: 20Hz-20kHz
 Outputs: Stereo 6.3mm headphone jack,
 plus internal loudspeaker

TCXR (frequency) accuracy: ± 3 ppm 0-60C, as standard,
 ± 0.5 ppm to special order

PLL responses: Loop filter -3dB LF corner at 700Hz/1.5kHz
 (reclocking and jitter Jitter gain at any frequency less than 1dB
 measurement) Intrinsic jitter <700ps (fc=1.5kHz)
 <1.2ns (fc = 700Hz)

CARRIER ANALYZER:

fs measurement: Range: 28kHz to 52kHz
 Accuracy, internal: set by TCXR as above
 external (wrt ref): <1ppm for 1 week

Jitter/eye measurement: Self-calibrating, peak-to-peak measurement
 Range: > 100ns P-P
 Low frequency range (-3dB),
 normal mode: PLL fc (700Hz or 1.5kHz)
 absolute (ref) mode: to <50Hz
 Accuracy: < $\pm 10\%$ plus residual jitter
 Residual jitter: <800ps, fc=1.5kHz
 <1.5ns at fc=700Hz

Carrier amplitude measurement: Auto-ranging with HF true-peak detector
 Range: 0 - greater than 20V P-P
 Frequency range, -3dB,
 wide-band: 300Hz-40MHz
 low-band: 28Hz-20kHz
 Accuracy: < greater of $\pm 5\%$ and ± 40 mV

Carrier phase measurement: Range: $\pm 180^\circ$ (50%)
 Accuracy: better than $\pm 1.4^\circ$ (0.39%)

DATA ANALYZER:

Audio meter readout: Range, dBFS mode: -138dBFS to 0dBFS
Accuracy, dBFS mode:
 ± 0.01 dB above -90dBFS
 ± 0.1 dB above -105dBFS
 ± 0.5 dB above -120dBFS
Accuracy, HEX mode: absolute

SIGNAL GENERATOR/EDITOR:

Sync sources: AES11, Wordclock, Int (44.1kHz, 48kHz),
AES3/IEC958 DI (in-line editing mode)

Generator 1:

Audio function: Sine, square, triangle, ramp, pulse (with
programmable polarity and duration), Jtest
(cable-jitter stimulator)

Frequency: At fs=48kHz: DC, 250, 500, 750, 1k, 1k5, 2k,
2k5, 3k, 4k, 5k, 6k, 7k, 8k, 9k, 10k, 11k, 12k,
13k, 14k, 15k, 16k, 17k, 18k, 19k, 20k, 21k,
22k, 23k, 24k;
Frequencies scaled for other sampling rates

Amplitude: 0dBFS, -1, -2, -3, -4, -5, -6, -7, -8, -9, -10, -
11, -12, -13, -14, -15, -16, -18, -20, -24, -30,
-36, -42, -48, -50, -60, -70, -80, -90, -100, -
110, -120dBFS
(-14dBFS, = +4dBu for +18dBu=0dBFS)

Audio word-length: 12-bits to 24-bits in 1-bit steps

Channel Status: All fields can be explicitly generated/edited;
entire Channel Status pattern can be
captured from a reference source

Validity: Can be set to 0 (valid) or 1 (invalid)

A and B sub-frames: Split mode: Each channel may have different
audio function, frequency, amplitude, word-
length, Channel Status and Valid bit if
required

In-line edit mode: Audio, Channel Status and Valid bit may be independently edited or passed through

Generator 2:

Audio function: fs/4 sine, Jtest (cable-jitter stimulator), PRS (pseudo-random sequence, white noise);

Channel Status: Professional or Consumer, with correct fs and CRC

Jitter generation: Function: PRS-H (wide-band noise), PRS-L (audio-band noise), fs/8 sine, long-cable-simulation
Amplitude P-P: 0 to $3UI/8$ in $UI/8$ steps, (20.3ns steps at fs=48kHz); $UI/12$ steps at fs=32kHz
Amplitude accuracy: $\pm 10\%$

In-line edit mode: Jitter generation can be added to the incoming signal

CHANNEL CHECKER:

Monitors any AES3 transmission medium and logs any bit-errors against time

Sub-frames tested: A, B or both
Word-length: 16, 20 or 24-bit
Time-stamp range: 1000 hours

WATCHDOG:

Monitors a selectable range of carrier and data parameters and logs fluctuations

Selectable parameters: Carrier fail/relock, biphasic/parity error, eye-width<50%, incorrect block-length/carrier phase, audio-band detection on carrier, fs (sampling-rate) base-value and accuracy, Audio-bit activity (main, low-main and auxiliary), Valid bit, User bit activity, Channel Status: Professional/Consumer, emphasis, sub-frame match, CRC errors, copy-protection, channel mode;
Time-stamp range: 1000 hours

MICROSCOPE:

Digital gain for examination of low-level signals, dithering schemes etc.

Gain range:	0 to +90dB in 6dB steps
Input word-length mask:	16, 20 or 24-bit
Clipping indicators:	A and B sub-frames independently

RESULTS LOG:

Non-volatile log capacity:	4095 bytes
Logged functions:	Carrier tests, Data tests, Test Sequences, Generator/editor settings, Channel Check, Watchdog
Printing/uploading:	Prints to serial printer during measurements or later from non-volatile log, or uploads to host PC
Printer baud rates:	1200, 2400, 4800, 9600

HOST SUPPORT:

Host PC requirement:	386 CPU or better, spare RS-232 serial port, MS-Windows 3.1 or above
Support programs:	MS-Windows: Test-Sequence Compiler/downloader, Results Log uploader/editor; MS-DOS: Firmware loader, Macro processor

Appendix A - Introduction to the AES3 interface

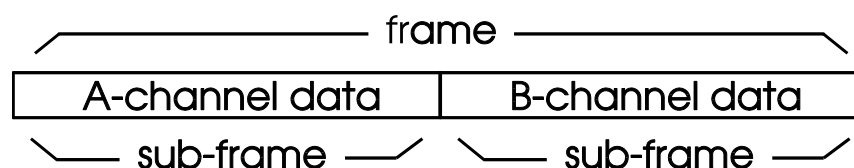
This section provides a brief introduction to the AES3 and IEC958 (professional and consumer) interface standards. For a full and detailed description, the reader is referred to the standards documents themselves, as listed in appendix D.

The later parts of this section describe problems which often arise in practical applications of the interface. Section 8 discusses ways in which the DSA-1 can be used in diagnosis of equipment and interfacing problems.

1. Carrier format

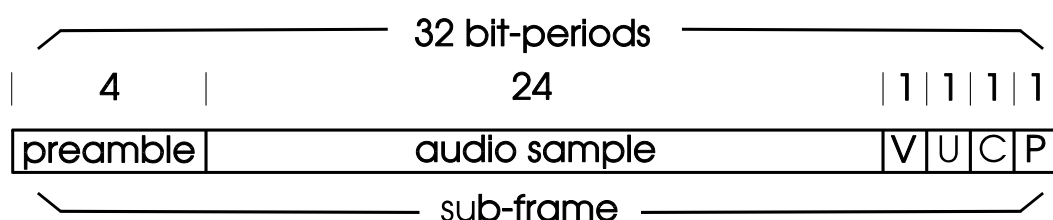
The AES3 interface carries two audio channels plus their associated status information on a single serial carrier. The interface is designed to operate 'point-to-point', i.e. each driver should be connected to no more than one receiver (although the possibility for drivers to feed multiple receivers was allowed in early versions of the AES3 standard).

The serial data for the two channels is interleaved in the carrier; first the A-channel data and status are transmitted, then the B-channel. Each channel-period is called a 'sub-frame' and the complete sample of A-channel and B-channel is a 'frame':

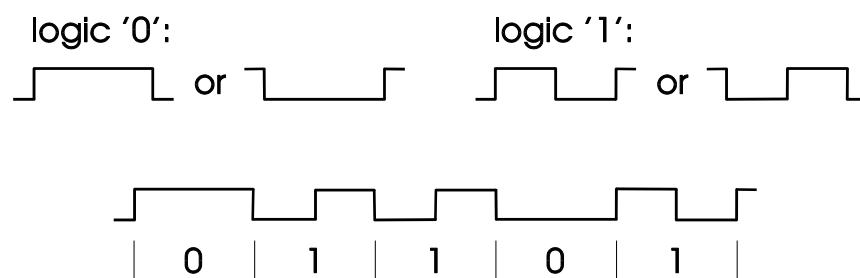


The serial data for each channel divided into 32 bit-periods, as follows:

- (First) 4 bit-periods for a preamble;
- 24 bit-periods for an audio sample;
- 1 bit-period for a 'Valid' bit;
- 1 bit-period for a 'User' bit;
- 1 bit-period for a 'Channel Status' bit;
- (Last) 1 bit-period for a 'Parity' bit;



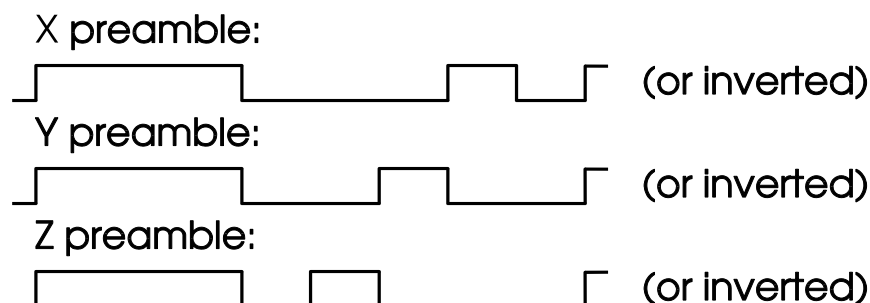
The binary data is 'biphase mark' encoded, i.e. each binary data bit is sent as two 'cells', with each bit-period bounded by low-to-high or high-to-low transitions. If the bit is a '1', another transition is put in the middle of the bit-period:



This coding scheme is often regarded as a way of 'combining clock and data' on a single carrier. An 'NRZ' serial data scheme (such as SDIF-2) has only one 'cell' per bit, so a long series of 0s or 1s would be impossible to recover since it would not be possible to tell where one bit ends and the next starts. SDIF-2 solves this problem by sending an extra clock signal on a separate cable (in the case of SDIF-2 a Wordclock, which is generally multiplied in a PLL in the receiver to derive a bit-reading clock). This has two drawbacks; an extra cable is required, and if the timing relationship between clock and data is lost (for example by use of unequal cable lengths for clock and data), the data may be incorrectly recovered. In contrast, the AES3 carrier needs only one cable for two channels and is easy to decode since transitions are guaranteed to be frequent.

A 'preamble' allows the start of the sub-frame to be picked out of what would otherwise be an endless stream of biphasic-mark 1s and 0s, because it violates the biphasic-mark rule: It doesn't have a transition at least every bit-period. Actually a different preamble pattern is transmitted for the A and B sub-frames so that the channels can be distinguished. These are the 'X' and 'Y' preambles respectively.

Only one Channel Status bit per channel per sample-period is carried on the interface, but these accumulate into a complex pattern over cycles of 192 sample periods, thus the entire Channel Status pattern is transacted every few milliseconds. In order to recognise the start of the 192-bit cycle, a third preamble type ('Z') is substituted for 'X' every 192 sample periods. The occurrence of 'Z' instead of 'X' precedes the first frame of the Channel Status cycle.



The primary audio sampling rates supported by AES3 are 32kHz, 44.1kHz and 48kHz. Close variants (for example 44.056kHz and 47.997kHz) also exist, often to accommodate NTSC video drop-frame/non-drop-frame variations. Some special application-specific rates also occur, such as 37.8kHz for CD-I. Lower rates are

beginning to appear for multi-media use. The bit-rate for AES3 interfaces is 64 times the sampling rate (64fs), i.e. 3.072MHz for fs=48kHz. The biphase-mark cell-rate is 128fs, i.e. 6.144MHz for fs=48kHz. The biphase-mark cell period (about 162ns for fs=48kHz) is referred to as a Unit Interval (or UI) since it represents the quantum pulse duration. In order that jitter and eye-width parameters can be scaled according to sampling rate, it is becoming common to refer to them in terms of UI rather than nanoseconds (ns).

2. Data format

The audio data word is transmitted least-significant-bit first, in two's-complement PCM format, and can be up to 24-bits in length. Where shorter word-lengths are used, these must be msb-aligned in the 24-bit field, and the preceding unused bits should usually be transmitted as '0'. Some alternative uses for the least significant four audio bits (the 'auxiliary bits') have been proposed, including their use for low-quality PCM or data-compressed channels. As yet, these are rarely encountered.

The Valid bit applies to the immediately preceding sample. It is sometimes interpreted as asserting whether or not the sample is suitable for conversion to analogue, sometimes whether or not any corrective interpolation has been applied to the signal in preceding equipment; some transports assert 'Invalid' when they are paused or stopped. A logic '0' is Valid, a logic '1' Invalid.

The User bit was originally just that - a bit whose use was not defined in the interface specification which was available to equipment or system builders to use as they wished. However, it soon became adopted for various 'standard' application-specific uses such as the transmission of CD sub-codes, DAT start-IDs etc. There is also an AES standard (AES18) applying to the proposed use of the User bit for imparting various types of data.

The 192-bit Channel Status cycle is notionally divided into 24 bytes, which are in turn subdivided into functional 'fields' of one or more bits each. The field definitions are different in the consumer and professional standards. The final byte of the professional Channel Status is a CRC (cyclic-redundancy-check-code) which allows a high probability of detecting a Channel Status error. For details of the functionality of the Channel Status fields, refer to the interface standards as listed in appendix D. Alternatively, the DSA-1's Generator Channel Status sub-menu 4.5. can be used to explore the possible settings.

The Parity bit is set to ensure that the sum of the audio, Valid, User and Channel Status bits in each sub-frame is always even. This is convenient for transmission hardware, since the biphase-mark encoding means that, with an even parity bit, the last data cell is always the same so that preambles are always transmitted in the same inversion.

3. Media format

At the present time, there are four principal media in use for the AES3 format:

Most professional installations currently use balanced, 110 Ω cables terminated in XLR connectors. The transmitter is usually transformer-coupled and source-terminated, and the loaded voltage is usually around 4V P-P since RS422 drivers are most often used. Receivers are terminated and nearly always transformer-coupled. Some de-facto conventions exist for grouping multiple balanced pairs on multiway connectors. Maximum transmission range is specified as 100m although in practice this can be reliably exceeded with suitable cables, or can be impossible to achieve with unsuitable types.

Consumer use is usually over quite short distances (the specified maximum is 10m), for which 75 Ω coaxial cable terminated with RCA (phono) plugs is normally used. Transmitter and receiver are terminated with 75 Ω , and the loaded voltage is 0.5V P-P. It is usual for only the transmitter to be transformer-coupled. The TOSLINK plastic optical system is also commonly used for consumer applications.

A newly-proposed format is AES3-ID, which seeks to take advantage of the potential similarity between an AES3 carrier and an analogue composite video signal. This format uses a 75 Ω coaxial video-type cable with BNC connectors. The transmitter and receiver are both terminated and transformer coupled. The loaded voltage is 1V P-P. Thus the level, bandwidth, cable and connector of AES3-ID are similar to analogue video; this allows easy use to be made of existing video cabling, patchfields, routers etc. The high bandwidth of the cable also reduces cable-induced jitter effects and increases transmission range to over 1000m.

4. Overview

In summary, we have a convenient two-channel carrier which is applicable to a variety of carrier media for different applications. It is reliably and simply decoded since it carries an embedded clock. Up to 24 audio bits per channel can be transmitted, plus Valid, User, Channel Status and Parity bits to enhance understanding of the signal context between equipment. The carrier is polarity-insensitive (since its information is embodied in transitions rather than logic levels) and DC-free (and so can be interfaced via isolating pulse-transformers).

5. AES3 interfacing problems

The following section deals briefly with a few causes of AES3 interfacing problems. It will be apparent that these problems beset practically every aspect of the format; however, this should not necessarily be viewed as a criticism of the format per se, since it undoubtedly represents a great improvement on what was previously available and, in any case, is now so well established as to be unalterable. Most of the difficulties

catalogued below are not the fault of the interface standard, but rather have resulted from its misuse by equipment manufacturers and system installers. Its inherent flexibility has often allowed problems to be created by unsuitable modes of use.

5.1. Carrier and cabling problems

The carrier format and cabling media themselves are the root of many interconnection problems:

On the one hand, the biphasemark coding scheme is clearly advantageous in maintaining data integrity - this is apparent when comparing the reliability of long-haul AES3 transmission with non-clock-embedded systems such as SDIF-2. The reason is clear when one considers what little may be left of a data stream after passage down a long cable - the reduction in amplitude and high-frequency content can make it difficult to decode even when the clock is embedded, but it is usually impossible to marry such data with a separate clock which may have come by a different route.

On the other hand, it is common for an AES3 signal to be used not only as a data interface, but also as a clock reference for conversion devices: most D/A converters derive their conversion clocks from their AES3 data input, and AES11 (essentially an AES3 signal) is the preferred reference distribution standard. In this role the AES3 carrier has not been so successful, since cable-induced jitter effects often cause distortion at the converter device ('sampling jitter') where long or poor quality cabling is used. The jitter is a direct consequence of the interaction between the high-frequency losses of the cabling and the clock/data embedding of the biphasemark coding scheme.

This effect is illustrated in figure A.1. below. Two alternative segments of transmitted biphasemark encoded data are shown, one carrying the binary sequence '000' and the other carrying '010'. The resultant cable voltage is shown for a long or poor-quality cable - the high-frequency losses of the cable result in the familiar 'charging/discharging' pattern at the receiver input. On examining the recovered data pattern after 'slicing' by the receiver's comparator, it can be seen that the timing of the latter two transitions in the '1' symbol has been displaced with respect to the 'correct' position which occurs when the symbol is a '0'.

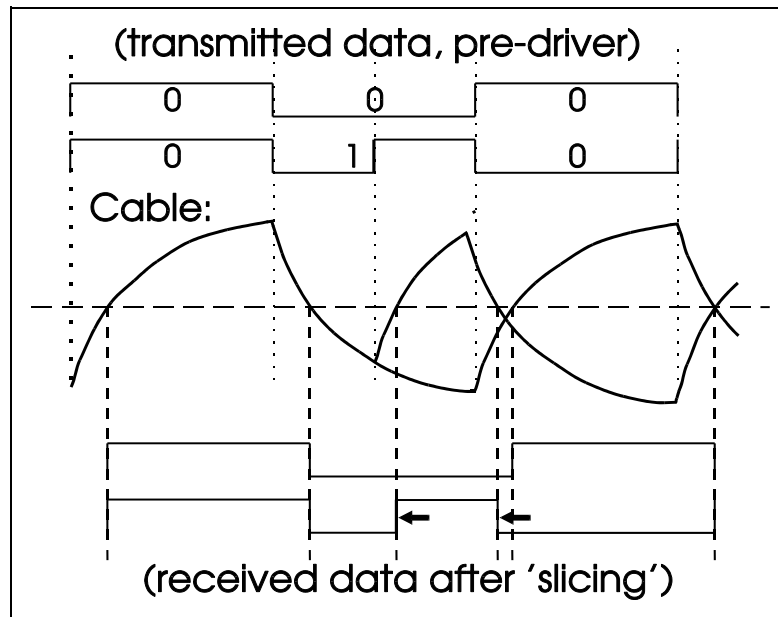


Figure A.1. - Data Jitter mechanism

This effect happens as a result of different pulse-lengths being present in the biphasemark data - if the data segment had carried '111', the recovered transition timing would have been 'correct', just as for '000'. If one imagines these three bits forming part of the audio sample in an AES3 signal, and alternating between '000' and '010' on successive samples, one can understand how a 'jittering' clock is recovered.

The complex dependence of data jitter on the data content itself can produce strange modulation effects if data jitter from a cable is allowed to cause sampling jitter in a converter device. Apart from the usual distortion products (which are aharmonic and unpleasant-sounding) these include 'crosstalk' between the two digital channels, since the conversion clock is shared by (and derived from) both sub-frames. As if this were not shocking enough, the crosstalk effect usually gets worse as the hostile channel gets *quieter*, because the coherence of audio frequency in the data pattern increases for smaller sample values when they are represented in twos-complement form. It is also possible for Channel Status and User data activity to modulate the converted audio in the same way.

The effect is variously referred to as 'data jitter', 'cable-induced jitter', 'intersymbol interference' or 'pulse smearing'. Data jitter (and interface jitter in general) should not be a problem so long as the data can still be recovered intact; the fault lies squarely with the manufacturers of conversion equipment, which should filter out the incoming jitter before the derived clock is applied to the converter device. However, this is a non-trivial analogue design problem which most converter manufacturers simply don't tackle. This required filtering is also called 'jitter attenuation' and is an important parameter in conversion equipment.

Interestingly, although AES11 was devised to provide a better method of reference distribution than Wordclocks which had been used until that point, this is seldom the case since Wordclocks, even if their pulse-shape is terribly distorted by long and lossy cables, can be received without intersymbol interference because they carry a simple square-wave rather than a varying data pattern. If the AES3 or AES11 signal providing the timing reference were 'black', i.e. containing '0' in all data and status bits, then only minimal data jitter could result because the data pattern would be 'static'. However, this never happens in practice: D/A converters normally extract their reference from the audio-bearing input, and AES11 references usually have active Channel Status identifying them as a reference, and sometimes carry a line-up tone as well.

There is an increasing trend towards the use of asynchronous sampling-rate converters in AES3 input circuits of conversion equipment, in order to implement jitter rejection more conveniently in the digital domain. However as yet, the signal-path quality of available devices precludes their use in top quality converters.

Whilst we have the situation where almost all available conversion equipment has unsatisfactory jitter attenuation, it is important to select cabling which will keep data jitter to a minimum. Generally, this means cable with minimum high-frequency losses, which usually comes down to capacitance (and resistance) per unit length in AES3 applications. Coaxial cables are usually very good, but it takes some determination to use these with equipment with XLR connectors directly, although matching transformer-barrels are available but quite expensive. Twin-axial cables such as were used for mini-computer networking probably provide the best performance for long-haul balanced AES3, but these are expensive, difficult to terminate, thick and astonishingly inflexible. Purpose-designed balanced AES3 cables in single-bearer and multiple-bearer configurations give adequate performance in most cases and are compact and not too expensive. In consumer installations, run lengths are generally too short for cabling effects to be considerable, although this view is not usually supported by consumer cable vendors.

5.2. Audio word-length problems

The AES3 interface supports up to 24-bits in the audio word. This is sometimes criticised as being insufficiently 'future-proof' but it is certainly hard to see it as a practical dynamic range limitation in interfaces between equipment.

However, a problem which does frequently occur is another example of the AES3 interface being, perhaps, too flexible: Many of the older digital audio interfacing standards had an inherently fixed audio word-length, usually 16 bits - all equipment using these interfaces supported the same audio word-length. In AES3 interfacing, the word-length can be anything from 16 to 24 bits, and problems often arise from different expectations between the transmitting and receiving equipment as to the word-length to be used. These difficulties theoretically need not occur, since the Channel Status specification allows the source to inform the destination of the precise word-length in use. However, most sourcing equipment does not generate this Channel Status, which

surprisingly turns out not to matter since most receiving equipment does not examine it anyway.

Where the receiving equipment is capable of accepting a greater word-length than is being sent by the transmitting equipment, no problem results because no audio information is lost.

When the receiving equipment is only capable of handling a shorter word-length than is being sent by the source, however, problems occur because the data is truncated to the shorter word-length on entering the receiving equipment, causing unpleasant aharmonic distortion products which cannot be subsequently repaired. It is normally beyond the capability of the receiving equipment to re-dither or noise-shape the incoming signal to the shorter word-length prior to truncation, as should be done, which is presumably why it does not bother to examine the input Channel Status. It is very common to find outboard 20-bit A/D converters being used with 16-bit recorders without re-dithering or noise-shaping being used to reduce the word-length. Even if the converter is capable of generating a reduced word-length output, the 20-bit setting is frequently used instead since it is assumed to offer better sound quality. In fact the above scheme will normally produce a worse product than the recorder's own 16-bit converters, although the resulting 'brightness' is often not recognised as distortion.

5.3. Ancillary data problems

Difficulties also occur through misuse or non-operation of the various ancillary bits.

The Valid bit may cause strange behaviour over an interface because its meaning has been historically vague, as described above. One possible difficulty occurs when the source equipment flags a sample 'Invalid' to indicate that an uncorrectable media error has been concealed - this is quite common behaviour in CD players and other transports. The receiving equipment is very likely to worsen the repair job done by the transport's interpolator by itself holding the previous sample; however, this would actually be preferable to the most likely response, which would be muting of the sample, probably causing a sound similar to the original uncorrected error.

The main cause of equipment incompatibility is Channel Status. This has largely been caused by the freedom of equipment manufacturers to misinterpret the standard or to settle on their own implementation subset. In general, a successful philosophy has been for transmitting devices to be precise in their Channel Status implementation, and for receiving devices to be tolerant in their interpretation. All too frequently however, AES3 outputs have had little or no Channel Status implemented, whereas sometimes receiving devices have been draconian in muting the audio in response to any partial or incorrect Channel Status implementation, or to momentary CRC errors (CRC errors are intended to apply only to the Channel Status, and have no implication for the integrity of the audio). Consumer receivers have historically been very fussy about the Channel Status implementation which they require, partly through non-pragmatic

design and partly through the constraints placed upon them by various copy-management systems such as SCMS, as described below.

'Transparency' to the Validity, User and Channel Status bits is becoming an important issue as implementation levels rise. For example, most older AES3 digital-to-digital equipment produced its own Channel Status output pattern (usually only a partial implementation) which was unaffected by the incoming Channel Status, or was affected only by one or two input fields. In general, incoming Channel Status was not passed to the output, thus important status information about the signal, appended by the sourcing equipment, was lost. Blocking of User bit data is equally problematic, for example CD sub-code or DAT start-ID information may be destroyed. Channel Status transparency is also an important issue in preserving correct operation of copyright protection schemes, e.g. SCMS.

Quasi-transparency to ancillary data also presents a problem: Some equipment which does pass Channel Status, User and Valid bits introduces a change in time-alignment of the audio and status data on the way. This might cause a single-sample 'Invalidity' indication to result in a subsequent equipment 'sample-holding' over the wrong sample, or perhaps Channel Status timecode may become misaligned with audio. In the case of Channel Status, partially-transparent equipment must modify some fields whilst passing on the others intact, and so must generate its own CRC at the end of the Channel Status block. If such equipment receives a Channel Status CRC error; it must generate its own CRC error deliberately to warn following equipment to disregard the parts of the Channel Status which it has passed through. This may be difficult to do in some systems.

The issue of ancillary-data transparency is badly handled in current equipment, but the interface standards are not always clear as to what behaviour is required, so we may expect problems in these areas to continue or worsen.

Finally it should be noted that, surprisingly, the Parity bit is generally of little use in detecting the onset of interface failure. This is because the intersymbol interference effect usually means that one of the cells in the AES3 preamble is lost before a parity error occurs. In old receiver designs this was often serious because total lock of lock was the first indication of the onset of failure. In modern receivers the loss of a preamble cell is treated just like any other biphase violation, including a parity error. It is better for receiving or analysis equipment to monitor all biphase errors rather than to look for parity errors specifically.

5.4. SCMS

SCMS (Serial Copy Management System) is a convention which has been adopted for consumer equipment to prevent pirate-copying by only allowing first-generation copying of original copyrighted material. The SCMS status of a signal passed between equipment is indicated by a combination of various bits in the Consumer-format Channel Status: These are the Copy Prohibit bit (byte 0, bit 4), the Category Code (byte 1, bits 0-6), and the Category Code Extension (byte 1, bit 7).

SCMS depends on the use of specially equipped recording machines which examine these bits on the incoming digital audio interface and act accordingly. For example, DAT recorders record two bits known as ID6 onto the recording medium according to the SCMS information passed from the source. These bits control the SCMS Channel Status transmitted when the tape is played back and control the further recording of the audio data on other SCMS machines. If the recorder recognises that the material has ID6 set to indicate that copyright exists and the material is not original, then a digital copy is prohibited. However the 'rules' for converting the Copy Prohibit bit and Category Code to recorded ID6 bits are rather complicated and this has led to confusion in the professional industry, especially when SCMS and pre-SCMS equipment are connected together.

Appendix B - DSA-1 Serial Data Protocol

The DSA-1 serial port has two basic modes of operation:

- i) For serial printing, sends ASCII text without handshaking at a baud rate set in the Setup sub-menu, 9.1. setting 9.
- ii) For special host-interfacing operations, including:
 - Firmware loading;
 - Serial Slave mode;
 - Test Sequence downloading;
 - Results Log uploading;
 - Alarms (Watchdog and Channel Check modes);

uses defined host-protocol at 9600 baud.

This appendix describes the protocols used for category (ii) modes, with the exception of Firmware loading.

There should be no need for most users to be concerned with the contents of this appendix, since application programs to operate the host modes are provided with the DSA-1. This section is intended for programmers who need to know how to operate the DSA-1's host modes from proprietary host equipment.

1. General

The baud rate is fixed at 9600, with no parity, 1 stop bit, 8 data bits.

Hexadecimal numbers are represented in this appendix as '0x..'; thus decimal 10 is shown as 0x0a. Checksums are such that data and checksum total zero. The following ASCII transfer constants are used:

STX	Start of Text	0x02
ETX	End of Text	0x03
ACK	Acknowledge	0x06
NAK	Negative Acknowledge	0x15

2. Wakeup exchange

In all the following modes, the wakeup exchange is always transacted first. This allows the host to ascertain that communication with the DSA-1 is working, and to learn the hardware and firmware versions of the DSA-1 connected.

The wakeup exchange is as follows:

HOST**DSA-1**

STX
0x12
(transfer type, see below)
0xFF

ACK/NAK

[HOST repeats wakeup if nothing or NAK received, up to maximum retries]

STX
(hardware ver, ASCII)
(firmware ver hi, ASCII)
(firmware ver mid, ASCII)
(firmware ver low, ASCII)
(checksum, all since STX)

[HOST and DSA-1 now enter selected mode, or else HOST terminates with:]

STX
0x00
0x00

[DSA-1 now aborts mode]

The latest hardware version is 'C'; the latest firmware version is '200'.

3. Serial Slave mode

This mode allows the host to control the 'keyboard' of the DSA-1 and to read the LCD display and LEDs. This mode is used by the Macro Processor PC software. Note that Serial Slave mode can be instated in the DSA-1 in any part of its menu system EXCEPT Host Mode (sub-menu 9.5.).

Transfer type is set in the wakeup exchange as 0x03. When the wakeup is achieved, the following protocols are available:

3.a) Keypresses to DSA-1

The HOST sends a key code between delimiters, the DSA-1 echoes this back:

HOST

```
STX
0x3c
(key code)
0x3e
```

DSA-1

```
STX
0x3c
(key code)
0x3e
```

Possible key codes are:

```
0x0f:    dummy
0x0e:    (DSA-1 on/off button - no effect)
0x0d:    <
0x0c:    >
0x0b:    ENTER/NEXT
0x0a:    EXIT
0x09:    EDIT/CLEAR
0x08:    LS ON/OFF
0x07:    LS MODE
0x06:    VOL UP
0x05:    VOL DN
0x04:    DI
0x03:    TERM
0x02:    CHAN
0x01:    DO
0x00:    SYNC O/P
```

The availability of a 'dummy' key allows the HOST to poll the DSA-1 periodically to check that it is still connected and communicating successfully. Since there is no checksumming in the returned data this is recommended to maintain an accurate record of the state of the DSA-1.

3.b) LCD text from DSA-1

In Serial Slave mode the DSA-1 echoes its LCD data to the HOST. This data is sent as raw ASCII without delimiters, and with no checksumming. The notional cursor is assumed to move one space to the right after each character. Allowable characters are in the range 0x20-0x7F and 0xA0-0xFF. Some of the more unusual DSA-1 character codes may not translate directly to the character codes of the HOST.

Note that the current 'soft' character set which occupies 0x00-0x07 of the LCD display's character set are relocated to 0xC0-0xC7 in order not to conflict with link control characters (0xC0-0xC7 are Kanji characters in the LCD module and would thus not be used).

3.c) LCD control codes from DSA-1

These codes impart various LCD-control information to the HOST, allowing it to maintain an accurate image of the DSA-1's LCD display:

HOST	DSA-1
[Cursor position:]	STX 0x80 (x value, 0-15) (y value, 0-1)
ACK/NAK	
[Clear/Home:]	STX 0x81
ACK/NAK	
[Select new soft character set:]	STX 0x82 (set number, 0-1)
ACK/NAK	

The HOST must have a knowledge of the DSA-1 soft character sets if it is to emulate DSA-1 displays. Set 0 is used within the bit-activity sub-menu, set 1 is used generally and within the audio bargraph sub-menu.

3.d) LED states from DSA-1

Changes to the DSA-1 LED indicators are reported:

HOST**DSA-1**

STX
 0x83
 (bank number, 0-2)
 (led data, low byte)
 (led data, high byte)

ACK/NAK

Each bit set in the data byte reflects a lit LED on the DSA-1. The CHARGE indicator cannot be sent. The LED codes are as follows:

```
/* Bank 0 */
C_FAIL_LED      0x0001h;
C_ERROR_LED     0x0002h;
C_NFAIL_LED     0x0004h;
C_BLOCK_LED     0x0008h;
C ALOG_LED     0x0010h;
C_32_LED       0x0020h;
C_44056_LED    0x0040h;
C_441_LED      0x0080h;
C_48_LED       0x0100h;
C_OTHER_LED    0x0200h;
C_200PPM_LED   0x0400h;
C_50PPM_LED    0x0800h;
C_10PPM_LED    0x1000h;
C_2PPM_LED     0x2000h;
C_WRTREF_LED   0x4000h;
K_LS_ON_LED    0x8000h;
```

```
/* Bank 1 */
D_AUX_LED      0x0001h;
D_AUDIOLO_LED  0x0002h;
D_AUDIOHI_LED  0x0004h;
D_V_LED        0x0008h;
D_U_LED        0x0010h;
D_PRO_LED      0x0020h;
D_CONS_LED     0x0040h;
D_1550_LED     0x0080h;
D_J17_LED      0x0100h;
D_ABDIF_LED    0x0200h;
D_CRCERROR_LED 0x0400h;
D_COPYPRO_LED  0x0800h;
D_STEREO_LED   0x1000h;
D_2CH_LED      0x2000h;
D_MONO_LED     0x4000h;
K_ON_ON_LED    0x8000h;
```

```
/* Bank 2 */
K_LSM_CHAN_LED 0x0001h;
K_LSM_BOTH_LED 0x0002h;
K_LSM_LINE_LED 0x0004h;
K_DI_XLR_LED   0x0008h;
```

K_DI_COAX_LED	0x0010h;
K_DI_OPT_LED	0x0020h;
K_TERM_LOZ_LED	0x0040h;
K_TERM_HIZ_LED	0x0080h;
K_CHAN_A_LED	0x0100h;
K_CHAN_B_LED	0x0200h;
K_DO_LOOP_LED	0x0400h;
K_DO_RECLK_LED	0x0800h;
K_DO_FUNC_LED	0x1000h;
K_SYNC_CH_LED	0x2000h;
K_SYNC_BLK_LED	0x4000h;
K_SYNC_REF_LED	0x8000h;

3.e) Exit Serial Slave mode

To exit from Serial Slave mode, the usual exit message is sent by the HOST:

HOST

STX
0x00
0x00

DSA-1

[DSA-1 now exits Serial Slave mode]

3.f) Alarms

In Watchdog or Channel Check modes, serial Alarm codes may be sent out by the DSA-1. This can happen even if Serial Slave mode is active. Refer to section 6. below for details of the Alarm messages.

4. Test Sequence downloads

In this mode, the HOST sends up to four User Test Sequences to the DSA-1 for storage in non-volatile memory. Each Sequence is 991 bytes long, i.e. shorter Sequences must be zero-padded by the downloader. Any or all of the four possible Sequence blocks may be updated during the download exchange.

Note that Test Sequence download mode can only be instated in the DSA-1 if it is in Host Mode (sub-menu 9.5.).

The transfer type set in the wakeup exchange is 0x01. After wakeup is established, the following transaction occurs:

HOST

```
STX
(block no - 1 to 4)
(data 000)
..
(data 990)
(checksum, all data not including block no)
```

DSA-1

ACK if received OK, else NAK

[HOST repeats block if NAK received, up to maximum retries]

[Repeat for next block, until all required blocks have been transferred]

```
STX
0x00 [end of block]
0x00 [dummy checksum]
```

[DSA-1 now exits mode]

5. Results Log uploads

In this mode, the DSA-1 is prompted to transmit the contents of its Results Log to the HOST. In the version 2.0 firmware, the DSA-1 transmits a variable number of bytes according to how much of the Log area is in use.

Note that Results Log upload mode can only be instated in the DSA-1 if it is in Host Mode (sub-menu 9.5.).

The transfer type set in the wakeup exchange is 0x02. After wakeup is established, the following transaction occurs:

HOST	DSA-1
	STX
	(data byte 0)
	...
	(data byte n-1)
	ETX
	(checksum, all data + ETX)
	STX [end token block]
	0x00
	0x00
	[DSA-1 now exits mode]
ACK (optional)	
[or redo wakeup exchange if block was bad]	

6. Alarms (Watchdog and Channel Check modes)

The DSA-1 may send short 'alarm' messages while in Watchdog or Channel Check modes. These alarms will be transmitted, if enabled, whether Serial Slave mode is active or not, so any HOST program that communicates with the DSA-1 in Serial Slave mode must be prepared to handle these messages. Each alarm consists of the following:

DSA-1

STX
0x84
alarm identifier

where the identifier is as follows:

```
/* WATCHDOG, CARRIER */
```


'* CARRIER FAIL'	0x00	'Carrier Locked'	0x01
'Biphase Error '	0x02	[unallocated]	0x03
'Near-Fail '	0x04	[unallocated]	0x05
'Block Lgth Err'	0x06	[unallocated]	0x07
'Low-band Sense'	0x08	'Low-band Clear'	0x09
'fs = 32k0Hz '	0x0a	'fs <> 32k0Hz '	0x0b
'fs = 44k056Hz '	0x0c	'fs <> 44k056Hz '	0x0d
'fs = 44k1Hz '	0x0e	'fs <> 44k1Hz '	0x0f
'fs = 48k0Hz '	0x10	'fs <> 48k0Hz '	0x11
'fs = Other '	0x12	'fs <> Other '	0x13
'fs = +-200ppm '	0x14	'fs <> +-200ppm'	0x15
'fs = +-50ppm '	0x16	'fs <> +-50ppm '	0x17
'fs = +-10ppm '	0x18	'fs <> +-10ppm '	0x19
'fs = +-2ppm '	0x1a	'fs <> +-2ppm '	0x1b
'Phase Error '	0x1c	[unallocated]	0x1d
[unallocated]	0x1e	[unallocated]	0x1f
/* WATCHDOG, DATA */			
'Aux Aud Activ'	0x20	'Aux Aud Black'	0x21
'20b Aud Activ'	0x22	'20b Aud Black'	0x23
'Main Aud Activ'	0x24	'Main Aud Black'	0x25
'V=1, Invalid '	0x26	'V=0, Valid '	0x27
'User Data '	0x28	'No User Data '	0x29
'C-S:Profes'nal'	0x2a	[unallocated]	0x2b
'C-S:Consumer '	0x2c	[unallocated]	0x2d
'C-S:15/50 emph'	0x2e	'C-S:15/50 off '	0x2f
'C-S:J17 emph '	0x30	'C-S:J17 off '	0x31
'C-S:A<>B '	0x32	'C-S:A=B '	0x33
'C-S:CRC Error '	0x34	'C-S:CRC OK '	0x35
'C-S:Copy Prot '	0x36	'C-S:Copy Allow'	0x37
'C-S:Stereo '	0x38	'C-S:Not Stereo'	0x39
'C-S:2-Channel '	0x3a	'C-S:Not 2-Chan'	0x3b
'C-S:Mono '	0x3c	'C-S:Not Mono '	0x3d
[unallocated]	0x3e	[unallocated]	0x3f
/* CHANNEL CHECK */			
'Locked on seq '	0x40	[unallocated]	0x41
'Sequence Fail '	0x42	'Sequenc ReLock'	0x43
'* CARRIER FAIL'	0x44	'Carrier Locked'	0x45
'* GEN MODE ERR'	0x46	'Gen Mode OK '	0x47
'Sample Error A'	0x48	'Sample Error B'	0x49

7. Expansion of serial modes

Further serial modes may be initiated using new values of the mode type byte in the wakeup exchange.

Appendix C - System Test Sequence Listings

This appendix contains listings of the four System (built-in) Test Sequences. These are included as example source files with the Test Sequence Compiler software. You may find it useful to use one of these Test Sequences as a starting point in developing your own User Test Sequences.

Sequence 01: Global Test

This Sequence allows Consumer or Professional Channel Status, and is quite tolerant of all parameters:

```

HEADER('Global Test      ');

/*
Segment 1: tests carrier for lock, near-fail, block-length, and analogue
*/
SEGMENT( 1,'Carrier qual');

CLEAR_ERROR_COUNT;
SET_FAIL;

    TEST_UNLOCK_BIT; JUMP_IF_ZERO(@lock_ok);
    MESSAGE( 'No carrier lock '); SEGMENT_END;
@lock_ok:
    TEST_NFAIL_BIT; JUMP_IF_ZERO(@carrier_ok);
    MESSAGE('Carrier nr fail '); SEGMENT_END;
@carrier_ok:
    TEST_BLOCKERR_BIT; JUMP_IF_ZERO(@block_ok);
    MESSAGE('Block length err'); SEGMENT_END;
@block_ok:
    TEST ALOG_BIT; JUMP_IF_ZERO(@not_alog);
    MESSAGE('Analogue signal '); SEGMENT_END;
@not_alog:

SET_PASS;
SEGMENT_END;
/*****/

/*
Segment 2: tests for a standard sampling rate, within 1000ppm
*/
SEGMENT( 2,'fs frequency');

SET_PASS;

    MEAS_FS_BASE; JUMP_IF_GREATER_THAN(0,@not_48);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm480,@bad_48);
    SEGMENT_END;
@bad_48:
    SET_FAIL; MESSAGE('48kHz, >1000ppm '); SEGMENT_END;

@not_48:
    JUMP_IF_GREATER_THAN(1,@not_441);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm441,@bad_441);
    SEGMENT_END;
@bad_441:
    SET_FAIL; MESSAGE('44kHz, >1000ppm '); SEGMENT_END;

@not_441:
    JUMP_IF_GREATER_THAN(2,@not_440);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm440,@bad_440);
    SEGMENT_END;
@bad_440:

```

```

    SET_FAIL; MESSAGE('44056, >1000ppm '); SEGMENT_END;

@not_440:
    JUMP_IF_GREATER_THAN(3,@not_32);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm320,@bad_32);
    SEGMENT_END;
@bad_32:
    SET_FAIL; MESSAGE('32kHz, >1000ppm '); SEGMENT_END;

@not_32:
    SET_FAIL; MESSAGE('Non-standard fs '); SEGMENT_END;
/*****

/*
Segment 3: tests amplitude according to input medium
*/
SEGMENT( 3,'Amplitude ');

SET_PASS; TEST_OPT; /* do not test if optical input */
JUMP_IF_ZERO(@not_opt); SEGMENT_END;

@not_opt:
    TEST_COAX;
    JUMP_IF_ZERO(@not_coax);

    MEAS_DIF_AMPL; /* coax ampl > 200mv, <2V */
    JUMP_IF_GREATER_THAN(2V,@coax_tobig);
    JUMP_IF_GREATER_THAN(200mv,@coax_done);
    SET_FAIL; MESSAGE('Coax ampl <0.2V');
@coax_done:
    SEGMENT_END;
@coax_tobig:
    SET_FAIL; MESSAGE('Coax ampl >2.0V'); SEGMENT_END;

@not_coax:
    MEAS_DIF_AMPL; /* XLR ampl > 400mv, <7V */
    JUMP_IF_GREATER_THAN(7V,@xlr_tobig);
    JUMP_IF_GREATER_THAN(400mv,@xlr_done);
    SET_FAIL; MESSAGE('XLR ampl <0.4V');
@xlr_done:
    SEGMENT_END;
@xlr_tobig:
    SET_FAIL; MESSAGE('XLR ampl >7.0V'); SEGMENT_END;
/*****

/*
Segment 4: tests for fs jitter within 20ns p-p, data jitter within 40ns p-p
*/
SEGMENT( 4,'Jitter ');

SET_PASS;

    MEAS_FS_JITTER(30); /* test for 3s */
    JUMP_IF_LESS_THAN(20ns,@fs_ok);
    MESSAGE('fs jitter >20ns '); SET_FAIL; SEGMENT_END;
@fs_ok:
    MEAS_DATA_JITTER(30); /* test for 3s */
    JUMP_IF_LESS_THAN(40ns,@data_ok);
    MESSAGE('Data jittr >40ns'); SET_FAIL;
@data_ok:
    SEGMENT_END;
/*****

/*
Segment 5: tests for eye reduction > 40ns (runt reduction)
*/
SEGMENT( 5,'Eye width ');

```

```

SET_PASS;

    SELECT_EYE_AT_ZEROX;
    MEAS_EYE_NARROW(30); /* test for 3s */
    JUMP_IF_LESS_THAN(40ns,@int_ok);
    MESSAGE('Smearing >40ns '); SET_FAIL;
@int_ok:
    SEGMENT_END;
/*****

/*
Segment 6: tests for non-audio channel status
*/
SEGMENT( 6,'Non-audio  ');

SET_PASS;

    TEST_CSBYTE(0); AND(02H); JUMP_IF_ZERO(@cs_ok);
    MESSAGE('Non-audio CS  '); SET_FAIL;
@cs_ok:
    SEGMENT_END;
/*****

/*
Segment 7: tests for any parity or biphase errors during the test sequence
*/
SEGMENT( 7,'Biphase errs');

SET_PASS;

    MEAS_ERROR_COUNT; JUMP_IF_ZERO(@code_ok);
    MESSAGE('Coding errors  '); SET_FAIL;
@code_ok:
    SEGMENT_END;

END; /* End of sequence */

```

Sequence 02: Consumer Test

This Sequence requires Consumer Channel Status, and applies Consumer-type limits similar to IEC958:

```

HEADER( 'Consumer Test  ');

/*
Segment 1: tests carrier for lock, near-fail, block-length, and analogue
*/
SEGMENT( 1,'Carrier qual',2);

CLEAR_ERROR_COUNT;
SET_FAIL;

    TEST_UNLOCK_BIT; JUMP_IF_ZERO(@carr_ok);
    MESSAGE('No carrier lock '); SEGMENT_END;
@carr_ok:
    TEST_NFAIL_BIT; JUMP_IF_ZERO(@nfail_ok);
    MESSAGE('Carrier nr fail '); SEGMENT_END;
@nfail_ok:
    TEST_BLOCKERR_BIT; JUMP_IF_ZERO(@block_ok);
    MESSAGE('Block length err'); SEGMENT_END;
@block_ok:
    TEST ALOG_BIT; JUMP_IF_ZERO(@not_alog);
    MESSAGE('Analogue signal '); SEGMENT_END;
@not_alog:
    SET_PASS; SEGMENT_END;
/*****

```

```

/*
Segment 2: tests for consumer channel status
*/
SEGMENT( 2, 'Consumer CS ');

SET_PASS;

TEST_CS_BYTE(0); AND(01H); JUMP_IF_ZERO(@cs_cons);
MESSAGE('CS not consumer '); SET_FAIL;
@cs_cons:
SEGMENT_END;
/*****

/*
Segment 3: tests for a standard sampling rate, within 1000ppm
*/
SEGMENT( 3, 'fs frequency');

SET_PASS;

MEAS_FS_BASE; JUMP_IF_GREATER_THAN(0, @not_48);
MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm480, @bad_48);
SEGMENT_END;
@bad_48:
SET_FAIL; MESSAGE('48kHz, >1000ppm '); SEGMENT_END;

@not_48:
JUMP_IF_GREATER_THAN(1, @not_441);
MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm441, @bad_441);
SEGMENT_END;
@bad_441:
SET_FAIL; MESSAGE('44kHz, >1000ppm'); SEGMENT_END;

@not_441:
JUMP_IF_GREATER_THAN(2, @not_440);
MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm440, @bad_440);
SEGMENT_END;
@bad_440:
SET_FAIL; MESSAGE('44056, >1000ppm '); SEGMENT_END;

@not_440:
JUMP_IF_GREATER_THAN(3, @not_32);
MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(1000ppm320, @bad_32);
SEGMENT_END;
@bad_32:
SET_FAIL; MESSAGE('32kHz, >1000ppm '); SEGMENT_END;

@not_32:
SET_FAIL; MESSAGE('Non-standard fs '); SEGMENT_END;
/*****

/*
Segment 4: tests amplitude according to input medium
*/
SEGMENT( 4, 'Amplitude ');

SET_PASS; TEST_OPT; /* do not test if optical input */
JUMP_IF_ZERO(@not_opt); SEGMENT_END;

@not_opt:
TEST_COAX; /* coax ampl > 200mv, <2V */
JUMP_IF_ZERO(@not_coax);

MEAS_DIF_AMPL;
JUMP_IF_GREATER_THAN(2V, @coax_tobig); JUMP_IF_GREATER_THAN(200mv, @coax_done);
SET_FAIL; MESSAGE('Coax ampl <0.2V');
@coax_done:
SEGMENT_END;
@coax_tobig:
SET_FAIL; MESSAGE('Coax ampl >2.0V'); SEGMENT_END;

```

```

@not_coax:
    MEAS_DIF_AMPL; /* XLR ampl (> 400mv, <7V */
    JUMP_IF_GREATER_THAN(7V,@xlr_tobig); JUMP_IF_GREATER_THAN(400mv,@xlr_done);
    SET_FAIL; MESSAGE('XLR ampl <0.4V');
@xlr_done:
    SEGMENT_END;
@xlr_tobig:
    SET_FAIL; MESSAGE('XLR ampl >7.0V'); SEGMENT_END;
/*****

/*
Segment 5: tests for fs jitter within 20ns p-p, data jitter within 40ns p-p
*/
SEGMENT( 5,'Jitter      ');

SET_PASS;

    MEAS_FS_JITTER(3s); /* test for 3s */
    JUMP_IF_LESS_THAN(20ns,@fs_ok);
    MESSAGE('fs jitter >20ns '); SET_FAIL; SEGMENT_END;
@fs_ok:
    MEAS_DATA_JITTER(3s); /* test for 3s */
    JUMP_IF_LESS_THAN(40ns,@data_ok);
    MESSAGE('Data jittr >40ns'); SET_FAIL;
@data_ok:
    SEGMENT_END;
/*****

/*
Segment 6: tests for eye reduction > 40ns (runt reduction)
*/
SEGMENT( 6,'Eye width  ');

SET_PASS;

    SELECT_EYE_AT_ZEROX;
    MEAS_EYE_NARROW(3s); /* test for 3s */
    JUMP_IF_LESS_THAN(40ns,@int_ok);
    MESSAGE('Smearing >40ns '); SET_FAIL;
@int_ok:
    SEGMENT_END;
/*****

/*
Segment 7: tests for non-audio channel status
*/
SEGMENT( 7,'Non-audio  ');

SET_PASS;

    TEST_CSBYTE(0); AND(02H); JUMP_IF_ZERO(@cs_ok);
    MESSAGE('Non-audio CS  '); SET_FAIL;
@cs_ok:
    SEGMENT_END;
/*****

/*
Segment 8: tests for any parity or biphase errors during the test sequence
*/
SEGMENT( 8,'Biphase errs');

SET_PASS;

    MEAS_ERROR_COUNT; JUMP_IF_ZERO(@code_ok);
    MESSAGE('Coding errors  '); SET_FAIL;
@code_ok:
    SEGMENT_END;
/*****
END;

```

Sequence 03: General Professional Test

This Sequence requires Professional Channel Status and applies general professional limits similar to AES3-1992:

```

HEADER( 'General Pro Test');

/*
Segment 1: tests carrier for lock, near-fail, block-length, and analogue
*/
SEGMENT( 1, 'Carrier qual');
CLEAR_ERROR_COUNT;
SET_FAIL;

    TEST_UNLOCK_BIT; JUMP_IF_ZERO(@carr_ok);
    MESSAGE('No carrier lock '); SEGMENT_END;
@carr_ok:
    TEST_NFAIL_BIT; JUMP_IF_ZERO(@nfail_ok);
    MESSAGE('Carrier nr fail '); SEGMENT_END;
@nfail_ok:
    TEST_BLOCKERR_BIT; JUMP_IF_ZERO(@block_ok);
    MESSAGE('Block length err'); SEGMENT_END;
@block_ok:
    TEST ALOG_BIT; JUMP_IF_ZERO(@not_alog);
    MESSAGE('Analogue signal '); SEGMENT_END;
@not_alog:
    SET_PASS; SEGMENT_END;
/*****

/*
Segment 2: tests for professional channel status
*/
SEGMENT( 2, 'Pro CS      ');

SET_PASS;

    TEST_CS_BYTE(0); AND(01H); JUMP_IF_NOT_ZERO(@cs_pro);
    MESSAGE('CS is consumer '); SET_FAIL;
@cs_pro:
    SEGMENT_END;
/*****

/*
Segment 3: tests for a standard sampling rate, within 50ppm
*/
SEGMENT( 3, 'fs frequency');
SET_PASS;

    MEAS_FS_BASE; JUMP_IF_GREATER_THAN(0, @not_48);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(50ppm480, @bad_48);
    SEGMENT_END;
@bad_48:
    SET_FAIL; MESSAGE('48kHz, >50ppm  '); SEGMENT_END;

@not_48:
    JUMP_IF_GREATER_THAN(1, @not_441);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(50ppm441, @bad_441);
    SEGMENT_END;
@bad_441:
    SET_FAIL; MESSAGE('44kHz, >50ppm  '); SEGMENT_END;

@not_441:
    JUMP_IF_GREATER_THAN(2, @not_440);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(50ppm440, @bad_440);
    SEGMENT_END;
@bad_440:
    SET_FAIL; MESSAGE('44056, >50ppm  '); SEGMENT_END;

```

```

@not_440:
    JUMP_IF_GREATER_THAN(3,@not_32);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(50ppm320,@bad_32);
    SEGMENT_END;
@bad_32:
    SET_FAIL; MESSAGE('32kHz, >50ppm '); SEGMENT_END;

@not_32:
    SET_FAIL; MESSAGE('Non-standard fs '); SEGMENT_END;
/*****

/*
Segment 4: tests amplitude according to input medium
*/
SEGMENT( 4,'Amplitude ');

SET_PASS; TEST_OPT; /* do not test if optical input */
JUMP_IF_ZERO(@not_opt); SEGMENT_END;

@not_opt:
    TEST_COAX; /* coax ampl > 200mv, <2V */
    JUMP_IF_ZERO(@not_coax);

    MEAS_DIF_AMPL;
    JUMP_IF_GREATER_THAN(2V,@coax_tobig); JUMP_IF_GREATER_THAN(200mV,@coax_done);
    SET_FAIL; MESSAGE('Coax ampl <0.2V');
@coax_done:
    SEGMENT_END;
@coax_tobig:
    SET_FAIL; MESSAGE('Coax ampl >2.0V'); SEGMENT_END;

@not_coax:
    MEAS_DIF_AMPL; /* XLR ampl > 400mv, <7V */
    JUMP_IF_GREATER_THAN(7V,@xlr_tobig); JUMP_IF_GREATER_THAN(400mV,@xlr_done);
    SET_FAIL; MESSAGE('XLR ampl <0.4V');
@xlr_done:
    SEGMENT_END;
@xlr_tobig:
    SET_FAIL; MESSAGE('XLR ampl >7.0V'); SEGMENT_END;
/*****

/*
Segment 5: tests for fs jitter within 20ns p-p, data jitter within 20ns p-p
*/
SEGMENT( 5,'Jitter ');

SET_PASS;

    MEAS_FS_JITTER(3s); /* test for 3s */
    JUMP_IF_LESS_THAN(20ns,@fs_ok);
    MESSAGE('fs jitter >20ns '); SET_FAIL; SEGMENT_END;
@fs_ok:
    MEAS_DATA_JITTER(3s); /* test for 3s */
    JUMP_IF_LESS_THAN(20ns,@data_ok);
    MESSAGE('Data jittr >20ns'); SET_FAIL;
@data_ok:
    SEGMENT_END;
/*****

/*
Segment 6: tests for eye reduction > 30ns (runt reduction)
*/
SEGMENT( 6,'Eye width ');

SET_PASS;

    SELECT_EYE_AT_ZEROX;
    MEAS_EYE_NARROW(3s); /* test for 3s */
    JUMP_IF_LESS_THAN(30ns,@int_ok);
    MESSAGE('Smearing >30ns '); SET_FAIL;

```



```

@int_ok:
    SEGMENT_END;
/*****

/*
Segment 7: tests for various channel status peculiarities
*/
SEGMENT( 7,'Channel Stat');

SET_PASS;

    TEST_CSBYTE(0); AND(02H);                                /* non-audio? */
    JUMP_IF_ZERO(@audio_cs);
    MESSAGE('Non-audio CS '); SET_FAIL; SEGMENT_END;
@audio_cs:
    TEST_CSBYTE(0); AND(1CH);                                /* reserved emphasis code? */
    JUMP_IF_EQUAL( 08H, @emph_rsvd);
    JUMP_IF_EQUAL( 10H, @emph_rsvd);
    JUMP_IF_EQUAL( 14H, @emph_rsvd);
    JUMP_IF_EQUAL( 18H, @emph_rsvd);
    JUMP(@emph_ok);
@emph_rsvd:
    MESSAGE('Reserved emph CS'); SET_FAIL; SEGMENT_END;
@emph_ok:
    TEST_CRCERR_BIT;                                         /* CRC not correct or 0? */
    JUMP_IF_ZERO(@cs_ok);
    TEST_CSBYTE(23); JUMP_IF_ZERO(@cs_ok);
    MESSAGE('CRC <> 0 or true'); SET_FAIL;
@cs_ok:
    SEGMENT_END;
/*****

/*
Segment 8: tests for any parity or biphase errors during the test sequence
*/
SEGMENT( 8,'Biphase errs');

SET_PASS;

    MEAS_ERROR_COUNT; JUMP_IF_ZERO(@code_ok);
    MESSAGE('Coding errors '); SET_FAIL;
@code_ok:
    SEGMENT_END;
/*****

END;

```

Sequence 04: Strict Professional Test

This Sequence is recommended by Prism Sound for use in professional installations. It requires good electrical and timing margins, and a full Professional Channel Status implementation:

```

HEADER( 'Strict Pro Test ');

/*
Segment 1: tests carrier for lock, near-fail, block-length, and analogue
*/
SEGMENT(1,'Carrier qual');
    CLEAR_ERROR_COUNT; SET_FAIL;
    TEST_UNLOCK_BIT; JUMP_IF_EQUAL(0,@lock_ok);
    MESSAGE('No carrier lock ');
    SEGMENT_END;
@lock_ok:
    TEST_NFAIL_BIT; JUMP_IF_EQUAL(0,@carr_ok);

```

```

    MESSAGE('Carrier nr fail ');
    SEGMENT_END;
@carr_ok:
    TEST_BLOCKERR_BIT; JUMP_IF_EQUAL(0,@block_ok);
    MESSAGE('Block length err');
    SEGMENT_END;
@block_ok:
    TEST ALOG_BIT; JUMP_IF_EQUAL(0,@not_alog);
    MESSAGE('Analogue signal ');
    SEGMENT_END;
@not_alog:
    SET_PASS; SEGMENT_END;
/*****

/*
Segment 2: tests for professional channel status
*/
SEGMENT(2,'Pro CS      ');
    SET_PASS;
    TEST_CS_BYTE(0); AND(01H); JUMP_IF_NOT_EQUAL(0,@cs_prof);
    MESSAGE('CS is consumer ');
    SET_FAIL;
@cs_prof:
    SEGMENT_END;
/*****

/*
Segment 3: tests for a standard sampling rate, within 10ppm
*/
SEGMENT(3,'fs frequency');
    SET_PASS; MEAS_FS_BASE; JUMP_IF_GREATER_THAN(0,@not_48k);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(10ppm480,@bad_48k);
    SEGMENT_END; // good 48k
@bad_48k:
    SET_FAIL; MESSAGE('48kHz, >10ppm '); SEGMENT_END;
@not_48k:
    JUMP_IF_GREATER_THAN(1,@not_44k1);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(10ppm441,@bad_44k1);
    SEGMENT_END; // good 44k1
@bad_44k1:
    SET_FAIL; MESSAGE('44k1Hz, >10ppm '); SEGMENT_END;
@not_44k1:
    JUMP_IF_GREATER_THAN(2,@not_44k0);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(10ppm440,@bad_44k0);
    SEGMENT_END; // good 44056
@bad_44k0:
    SET_FAIL; MESSAGE('44056, >10ppm '); SEGMENT_END;
@not_44k0:
    JUMP_IF_GREATER_THAN(3,@not_32k);
    MEAS_FS_ERROR; AND(07FFFH); JUMP_IF_GREATER_THAN(10ppm320,@bad_32k);
    SEGMENT_END;
@bad_32k:
    SET_FAIL; MESSAGE('32kHz, >10ppm '); SEGMENT_END;
@not_32k:
    SET_FAIL; MESSAGE('Non-standard fs '); SEGMENT_END;
/*****

/*
Segment 4: tests amplitude according to input medium
*/
SEGMENT(4,'Amplitude ');
    SET_PASS; TEST_OPT; /* do not test if optical input */
    JUMP_IF_EQUAL(0,@not_opt);
    SEGMENT_END;
@not_opt:
    TEST_COAX; /* coax ampl > 200mv, <2V */
    JUMP_IF_EQUAL(0,@do_xlr);
    MEAS_DIF_AMPL;
    JUMP_IF_GREATER_THAN(2V,@coax_above);
    JUMP_IF_GREATER_THAN(200mV,@coax_ok);
    SET_FAIL; MESSAGE('Coax ampl <0.2V');

```

```

@coax_ok:
    SEGMENT_END;
@coax_above:
    SET_FAIL; MESSAGE('Coax ampl  >2.0V');
    SEGMENT_END;

@do_xlr:
    MEAS_DIF_AMPL;    /* XLR ampl , > 400mv, <7V*/
    JUMP_IF_GREATER_THAN(7V,@xlr_above);
    JUMP_IF_GREATER_THAN(400mV,@xlr_ok);
    /* too low */
    SET_FAIL; MESSAGE('XLR  ampl  <0.4V');
@xlr_ok:
    SEGMENT_END;
@xlr_above: /* too high */
    SET_FAIL; MESSAGE('XLR  ampl  >7.0V'); SEGMENT_END;
/*****

/*
Segment 5: tests for fs jitter within 5ns p-p, data jitter within 10ns p-p
*/
SEGMENT(5,'Jitter      ');
SET_PASS;

    MEAS_FS_JITTER( 3s );    /* test for 3s */
    JUMP_IF_LESS_THAN(5ns,@fs_ok);          /* 64ths of a ns */
    MESSAGE('fs jitter >5ns '); SET_FAIL;
    SEGMENT_END;
@fs_ok:
    MEAS_DATA_JITTER( 3s );    /* test for 3s */
    JUMP_IF_LESS_THAN(10ns,@data_ok);        /* 64ths of a ns */
    MESSAGE('Data jittr >10ns'); SET_FAIL;

@data_ok:
    SEGMENT_END;
/*****

/*
Segment 6: tests for eye reduction > 10ns (runt reduction)
*/
SEGMENT(6,'Eye width  ');

    SET_PASS;
    SELECT_EYE_AT_ZEROX;
    MEAS_EYE_NARROW( 3s );/* test for 3s */
    JUMP_IF_LESS_THAN(10ns,@int_ok);          /* 64ths of a ns */
    MESSAGE('Smearing >10ns '); SET_FAIL;

@int_ok:
    SEGMENT_END;
/*****

/*
Segment 7: tests for various channel status peculiarities
*/
SEGMENT(7,'Channel Stat');
SET_PASS;

    TEST_CSBYTE( 0 ); AND(02H);                /* non-audio? */
    JUMP_IF_EQUAL(0,@cs_audio);
    MESSAGE('Non-audio CS      '); SET_FAIL; SEGMENT_END;
@cs_audio:
    TEST_CSBYTE( 0 ); AND(1CH);                /* reserved emphasis code? */
    JUMP_IF_EQUAL( 08H, @rsvd_emph);
    JUMP_IF_EQUAL( 10H, @rsvd_emph);
    JUMP_IF_EQUAL( 14H, @rsvd_emph);
    JUMP_IF_EQUAL( 18H, @rsvd_emph);
    JUMP(@emph_ok);
@rsvd_emph:
    MESSAGE('Reserved emph CS'); SET_FAIL; SEGMENT_END;

```

```

@emph_ok:
    TEST_CRCERR_BIT;                      /* CRC not correct? */
    JUMP_IF_EQUAL(0,@crc_ok);
    MESSAGE('CRC error      '); SET_FAIL; SEGMENT_END;

@crc_ok:
    TEST_CSBYTE( 0 );                     /* src sampling frequency unlocked? */
    JUMP_IF_MASKED(20h , @src_lock );
    MESSAGE('Source unlocked '); SET_FAIL; SEGMENT_END;

@src_lock:
    TEST_CSBYTE( 0 ); AND(0C0H);           /* Channel Status fs <> actual fs? */
    JNE(080H, @cs_not48);                 /* CS 48k */
    MEAS_FS_BASE; JNE(0 ,@wrong_fs);
@cs_not48:
    JNE(040H, @cs_not44);                 /* CS 44k1 */
    MEAS_FS_BASE; JNE(1, @wrong_fs);
@cs_not44:
    JNE(0C0H, @fs_ok);                   /* CS 32k - skip if unknown CS */
    MEAS_FS_BASE; JUMP_IF_EQUAL( 3, @fs_ok);
@wrong_fs:
    MESSAGE('fs conflict      '); SET_FAIL; SEGMENT_END;
@fs_ok:
    TEST_CSBYTE( 1 );                     /* Reserved state of byte 1? */
    JUMP_IF_MASKED(031h, @notrsvd1);
    MESSAGE('Rsvd in CS byte1'); SET_FAIL; SEGMENT_END;
@notrsvd1:
    TEST_CSBYTE( 2 );                     /* Reserved state of byte 2? */
    JUMP_IF_MASKED(0C1h, @notrsvd2);
    MESSAGE('Rsvd in CS byte2'); SET_FAIL; SEGMENT_END;
@notrsvd2:
    TEST_CSBYTE( 3 );                     /* Reserved state of byte 3? */
    JUMP_IF_EQUAL(0, @notrsvd3);
    MESSAGE('Rsvd in CS byte3'); SET_FAIL; SEGMENT_END;
@notrsvd3:
    TEST_CSBYTE( 4 );                     /* Reserved state of byte 4? */
    JUMP_IF_MASKED(0FCh, @notrsvd4);
    MESSAGE('Rsvd in CS byte4'); SET_FAIL; SEGMENT_END;
@notrsvd4:
    TEST_CSBYTE( 5 );                     /* Reserved state of byte 5? */
    JUMP_IF_EQUAL(0 ,@notrsvd5);
    MESSAGE('Rsvd in CS byte5'); SET_FAIL; SEGMENT_END;
@notrsvd5:
    TEST_CSBYTE( 22 );                     /* Reserved state of byte 22? */
    JUMP_IF_MASKED(00Fh, @notrsvd22);
    MESSAGE('Rsvd in CSbyte22'); SET_FAIL;
@notrsvd22:
    SEGMENT_END;
/*****

/*
Segment 8: tests for any parity or biphase errors during the test sequence
*/
SEGMENT(8,'Biphase errs');
SET_PASS;
    MEAS_ERROR_COUNT; JUMP_IF_EQUAL(0,@code_ok);
    MESSAGE('Coding errors      '); SET_FAIL;
@code_ok:
    SEGMENT_END;
/*****/

END;

```

Appendix D - References

a) Interface standards documents:

- [1] AES3-1992 - 'Recommended Practice for Digital Audio Engineering - Serial Transmission Format for Two-Channel Linearly Represented Digital Audio Data' J. Audio Eng. Soc., Vol 40 No. 3, pp 147-165 (June 1992)

The 'AES/EBU' professional interface standard

- [2] IEC958 - 'Digital Audio Interface' International Electrotechnical Commission 1989

The consumer interface standard

- [3] IEC958 - 'Digital Audio Interface - Amendment 1' International Electrotechnical Commission 1992

Additions and amendments, including extensions to the allowed category codes, and SCMS (Serial Copy Management System)

- [4] IEC958 - 'Digital Audio Interface - Amendment 2' International Electrotechnical Commission 1995

Additions and amendments, including use of User data for DAT and DCC, and some new category codes

- [5] AES11-1991 - 'AES Recommended Practice for Digital Audio Engineering - Synchronization of Digital Audio Equipment in Studio Operations' J. Audio Eng. Soc., Vol. 39 No. 3, pp 155-162 (March 1991)

The use of 'AES/EBU' type signals as Reference Syncs

- [6] AES17-1991 - 'AES Standard Method for Digital Audio Engineering - Measurement of Digital Audio Equipment' J. Audio Eng. Soc., Vol. 39 No. 12, pp 961-975 (December 1991)

Suggested list of standard measurements for digital equipment

b) Additional information on Standards:

- [7] AES3-ID-1995 - 'AES information document for digital audio engineering - Transmission of AES3 formatted data by unbalanced coaxial cable' J. Audio Eng. Soc., Vol. 43 No. 10, pp 827-844 (October 1995)
- [8] AES Standards Committee News - 'Notice of proposed revisions to the AES3-1992 jitter specification' J. Audio Eng. Soc., Vol. 42 No. 12, pp 1026-1027 (December 1994)

c) Papers:

- [9] N.J.Dunn, Prism Sound, - 'Considerations for Interfacing Digital Audio Equipment to the Standards AES3, AES5, AES11' Published in 'Images of Audio', the Proceedings of the 10th International AES Conference, London, September 1991. pp 115-126
- [10] Julian Dunn and Ian Dennis, Prism Sound, - 'The Diagnosis and Solution of Jitter-Related Problems in Digital Audio Systems' Preprint 3868, presented at the 96th AES Convention, Amsterdam, February 1994.
- [11] D.G. Kirby - 'Twisted-Pair Cables for AES/EBU Digital Audio Signals' J. Audio Eng. Soc., Vol. 43 No. 3, pp 137- 146 (March 1995)
- [12] Julian Dunn, Prism Sound - 'Jitter and Digital Audio Performance Measurements' in 'Managing the Bit Budget', the proceedings of the AES UK Conference 16-17 May 1994, pp 148-166.
- [13] Julian Dunn, Barry McKibben, Roger Taylor and Chris Travis - 'Towards Common Specifications for Digital Audio Interface Jitter' Preprint 3705, presented at the 95th AES Convention, New York, October 1993

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