Prism*Sound*

Jitter Modulator

Operation Manual

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1. INTRODUCTION

Jitter can be the cause of total failure of digital audio equipment to communicate or a relatively small degradation in the audio signal quality produced at a converter. The JM-1 can generate jitter with various amplitudes and spectra. This can be used in the analysis and diagnosis of jitter-related problems related to individual pieces of equipment or studio installations.

2. SUPPLIED WITH YOUR JM-1

The JM-1 is supplied with a suitable mains lead and an operation manual (this document). This should be checked, on receipt, for any damage - which should be reported to your supplier as soon as possible.

3. SIMPLE JM-1 OPERATION

The JM-1 can be used straight away if you have two pieces of digital audio equipment that interface using a compatible digital audio format, such as SPDIF or AES3 (see later).

Set them up normally so that the source equipment, such as a CD player, feeds into the destination equipment, such as a DAC. When this is working replace the direct link by a connection via the JM-1.

The CD player should connect to one of the digital audio input connections on the JM-1. Power up the JM-1 and select the input being used using the push-button on the left side of the front panel. If that part of the connection is working then the `locked' indicator should illuminate.

Set the modulation source to low frequency noise by pressing the second push button until the `noise' and `LF' indicators are illuminated. Rotate the modulation level knob fully anti-clockwise (modulation zero) and set the range to x1ns by pressing the amplitude push-button until the indicator is not illuminated.

Connect the DAC to one of the digital audio outputs - they are all active simultaneously. The system should now work again, but this time you can apply a variable amount of jitter to the signal going to the DAC. Turn up the jitter amplitude to 50ns, using the rotary knob. If it still works then the jitter tolerance of the DAC is adequate for most normal levels of jitter. If you change the range to x10ns by pressing the amplitude push-button you should be able to get the unit to fail between 50 and 150ns of applied jitter. This is an indication of the margin of jitter tolerance the DAC has.

Another important test is for the effect of jitter on the audio quality of a DAC. The signal applied to the DAC, via the JM-1, may be a test signal, such as may be used for a THD+n (total harmonic distortion plus noise) measurement. Using this it is possible to use, for example, a THD+n meter on the DAC output, to assess how much the THD+n performance of the DAC is degraded by the interface jitter applied by the JM-1. (Note that the results of this test will depend on several factors, including the frequency and amplitude of the test signal - as well as the DAC architecture.)

4. FRONT PANEL CONTROLS AND INDICATORS

4.1. Digital Input Select

The digital audio input connector that is made active is selected by pressing the left-most push-button in the box marked digital input.

The selection made is shown by an illuminated \`XLR' or \`Coax' indicator.

If the active input is successfully decoded the \`locked' indicator is illuminated.

4.2. Modulation Source Select

The modulation sources are selected by pressing the push-button in the modulation source box. The sources are selected in turn by each successive depression of the push-button.

4.2.1. External (**Ext**)

This selects the external jitter modulation signal source. In most applications this would typically be a sine wave oscillator with an output level of 2V pk-pk.

4.2.2. Internal Square Wave (**SQ**)

This selects an internally generated filtered square wave of approximately 8kHz. (This signal is limited in harmonic content by a 20kHz first order low pass filter.)

4.2.3. Internal Wideband Noise (**Noise**)

This selects a high frequency filtered binomial (two-level) noise source generated by a pseudo-random binary sequence (PRBS) generator. The bandwidth of the signal is limited by a 1MHz first order low pass filter.

4.2.4. Internal Low Frequency Noise (**Noise LF**)

This selects a low frequency filtered binomial (two-level) noise source generated by a pseudo-random binary sequence (PRBS) generator. The bandwidth of the signal is limited by a 20kHz first order low pass filter.

4.3. External Source Calibrate

Adjacent to the modulation source control the `Ext Cal.' indicators can be used to set the level of the external modulation signal. The less-than or greater-than indicators illuminate if the signal is lower or higher than the required amplitude for calibrated operation. (That is when the front panel amplitude control calibrations are correct). When they are both illuminated the level is within 2% of the correct level. Note that these indicators are independent of the front panel control, and only need to be used to set the level of an un-calibrated external modulation source, such as a simple oscillator.

4.4. Jitter Amplitude Control

The rotary knob provides adjustment of jitter modulation amplitude. The calibrations are correct when using one of the internal modulation sources or a calibrated external source.

The amplitude range push-button selects between two ranges.

With the button indicator not illuminated the fine range (x1ns) is selected. In this range the scale corresponds with the peak to peak jitter amplitude in nanoseconds.

With the button indicator illuminated the coarse range (x10ns) is selected. In this range the scale needs to be multiplied by 10 to correspond with the peak to peak jitter amplitude in nanoseconds.

4.5. Sync Output Select

The right-most push-button selects the frequency of the synchronization output. This is for use with an oscilloscope and allows the operator to trigger from the un-jittered source signal so that jitter at the output of the JM-1, or at the output of equipment under test can be observed for the purposes of measuring the jitter gain or jitter transfer function of the following equipment.

With the button indicator not illuminated the sync pulse occurs once per frame (sample period), and selection of positive or negative-edge triggering will trigger an oscilloscope just before the start of either the first or second sub-frame.

When the button indicator is illuminated the sync frequency is 128xFS which will trigger the oscilloscope to catch every possible edge. This is useful when trace illumination is weak, or there is suspicion of transition asymmetry or jitter being locked to the frame or sub-frame rate in the device under test.

5. EXAMPLES - SOME TESTS USING THE JM-1

5.1. Interface receiver jitter tolerance

The jitter tolerance of a receiver is the amount of jitter that an interface signal can have before the receiver will fail to decode it reliably. The tolerance of a receiver varies with jitter frequency. Above a corner frequency the tolerance to jitter is normally flat or independent of frequency, while below that frequency it usually rises reciprocally with falling jitter frequency.

The high frequency jitter tolerance can be measured using the internal broadband noise modulation source. While monitoring the device under test for errors gradually increase the jitter modulation level (starting at the bottom of the x10ns range. A receiver will typically fail when stimulated with between 0.25UI (40ns) and 1UI (160ns) of high frequency jitter.

Low frequency jitter tolerance can be measured using a variable-frequency sine oscillator as the external modulation source. Spot measurements of jitter tolerance at 500Hz, 1kHz, 2kHz, 5kHz and 10kHz will give a good idea of the shape of the jitter tolerance curve.

Near the corner frequency of the jitter tolerance curve the tolerance may dip to significantly less than the high frequency tolerance level. This is a result of a large phase error in the data recovery phase-locked-loop, at that frequency. The corner frequency can be estimated by extrapolating the intersection between the LF tolerance curve and the HF tolerance level. Measurements at that frequency may reveal poor PLL phase margin.

5.2. Interface receiver to transmitter jitter transfer function

Equipment with an output that is `locked' to a digital audio input will pass through a degree of jitter from the input to the output. The relation between input and output jitter can often be approximated, except at low jitter levels, as a linear transfer function. This will have the character of a low pass filter.

The jitter transfer function can be measured using a variable frequency sine oscillator as the external modulation source to the JM-1. The output jitter of the device under test can be measured using an oscilloscope triggered from the JM-1 sync output and looking at the data transitions in the digital audio output of the device under test.

Set the JM-1 to provide 25ns of jitter on the x1ns range. Set the external oscillator to the calibrated level (2V pk-pk) using the cal function of the JM-1, if necessary, and feed the JM-1 digital audio input from a known low-jitter source with a short cable.

Measure the output jitter of the DUT (using the oscilloscope, as described above) and sweep the oscillator from the lowest frequency upwards. The output jitter should remain constant until the corner frequency of the jitter transfer function is reached. At this point the output jitter may rise to a maximum - the jitter peaking level - before falling with increasing frequency. This response may be plotted but the key feature is the corner frequency and the peaking level at that frequency.

In performing this measurement you may wish to note any aliasing that may occur as the jitter frequency approaches the sampling frequency and twice the sampling frequency. This is not a serious problem but might indicate that the equipment is using only transitions within the data preambles for synchronization. This will allow the equipment to be relatively insensitive to jitter in the modulated data part of the waveform. (There is a signal that can be used to reveal the data-jitter susceptibility, J-test, and this is described in reference 6 and can be generated by the Prism Sound DScope).

Please note that it is possible to use a weighted jitter meter, such as the Prism Sound DSA-1 for the jitter amplitude measurement. In this case the weighting characteristics of the meter will need to be allowed for by measuring the input and output jitter at each spot frequency and taking the ratio of the two. If the corner frequency of the DUT is significantly below that of the meter then the measurement will not be very accurate. (The weighting characteristic of the DSA-1 is designed for consistent intrinsic jitter measurements where the user will not necessarily have access to the sync signal feeding the DUT)

5.3. Interface receiver jitter to sampling jitter rejection

The sampling jitter of a digital audio converter, such as an analogue to digital (ADC), digital to analogue (DAC), or sampling rate converter (SRC) can be influenced by jitter on the digital audio input it uses for synchronization (either input in the case of a resynchronizing sample rate converter).

This relation can be approximated as in the previous example by a jitter transfer function with the characteristic of a low-pass filter. The JM-1 can be used - in conjunction with a high dynamic range spectrum analyzer - to measure this jitter transfer function. A high level and high frequency stimulus signal is required, such as a tone peaking at 1dB below full scale and at a frequency of a quarter of the sample rate (12kHz). This stimulus may be applied by the digital input, in the case of a DAC or SRC, in which case it would be fed via the JM-1 from a low jitter digital audio signal generator such as a Prism Sound DScope. For an ADC the stimulus should be generated by a high quality analogue signal generator.

The spectrum of the output signal, be it digital or analogue, is then analyzed for modulation sidebands that vary with applied jitter level. The applied jitter is white (flat with frequency) so the shape of the sideband skirts reveals the corner frequency of the receiver jitter to sampling jitter transfer function.

6. SPECIFICATIONS

6.1. Modulator Characteristics

6.1.1. Modulation Sources

The internal sources are generated digitally with a clock that is locked to the incoming digital audio sampling frequency (FS). As a consequence the noise and square wave frequencies are quoted with respect to FS.

The modulation signals are low-pass filtered to restrict the maximum jitter slewing (frequency deviation). The filters are simple 1st order networks with approximate 3dB points as quoted.

The noise signals are filtered two-level pseudo-random binary sequences, and the square wave is designed to drift in phase with respect to the sample clock.

Source		Period	Clock rate	Bandwidth
External	(BNC i/p)		-	1MHz

Square Wave	- 8kHz	- 5.953/FS	FS*64/381	20kHz
Wideband Noise	19 bit MLS	- 6144/FS	FS*256/3	1MHz
Low Frequency Noise	11 bit MLS	- 1015.5/FS	FS*256/127	20kHz

MLS - Maximum length sequence FS - Sampling frequency

6.1.2. Jitter Amplitude

The jitter amplitude is quoted peak to peak, and for an external source the accuracy quoted assumes a sinusoidal source of amplitude 2V pk-pk and frequency less than 100kHz.

Range x1ns 0 - 50ns Accuracy 5% or 2ns, whichever is greater

Range x10ns 0 - 500ns Accuracy 5% or 10ns, whichever is greater

With high frequency or wideband modulation it is possible for large jitter amplitudes to produce `illegal' signals as the jitter change between data transitions may cause the removal of the transitions. This will occur only at levels far in excess of the tolerance of any digital audio receivers known at present.

6.1.3. Modulator Frequency Response

At frequencies above 100kHz the modulator frequency response is affected by the accuracy of the bandwidth limiting filter and the filtering effect as a result of the integration of the modulation signal by the modulator. The integration times for zero modulation are:

Range x1ns 40ns
Range x10ns 400ns

This integration time is modulated by the applied jitter signal itself, so at the maximum levels of the x10ns range the modulation signal will be integrated over a range between 150ns and 650ns. This integration time will attenuate the high frequency components of the jitter.

6.2. Intrinsic Jitter

With no modulation and a jitter free input the output jitter, measured input to output, is less than:

Range x1ns	2ns
Range x10ns	10ns

6.3. Sampling Frequency Range

The output will track the input and work correctly for digital audio sample rates between 48.5kHz and 31.5kHz.

6.4. Digital Audio Input

6.4.1. BNC Input

Termination	75 ohms
Sensitivity	100mV
Transformer isolation	

This input will accept signals conforming electrically to the unbalanced IEC958 [2] format (SPDIF) and the proposed coaxial AES-3ID format [9]. The incoming data is passed through but otherwise ignored.

6.4.2. XLR Input

Termination	110 ohms
Sensitivity	500mV
Transformer isolation	

This input will accept signals conforming electrically to the balanced IEC958 or AES-3 format [1]. The incoming data is passed through but otherwise ignored.

6.5. Digital Audio Outputs

When the unit is locked to an input the unit will pass through the incoming data. When the unit is unlocked the output will be indeterminate and cannot be used for test or measurement.

The outputs carry identical digital data and the channel status at every output will be that at the input. This may not conform with the data formats which the connectors are intended to support. Therefore the JM-1 <u>does not translate</u> between consumer and professional variants of the digital audio interface.

All the digital outputs are separately buffered and transformer isolated (except for 10nF capacitors between the cable screens and the equipment case).

6.5.1. RCA phono output (0.5V 75S)

This conforms electrically and mechanically with the unbalanced IEC958 standard.

6.5.2. Optical output

This conforms optically and mechanically with the EIAJ CP-340 [4] optical interface standard. This optical connector format is also known as TosLink.

6.5.3. BNC output (1.0V 75S)

This conforms electrically and mechanically with the proposed format for the long distance transmission of AES3 signals [9]. This format is intended to be compatible with video transmission circuits.

This output is also suitable for feeding an oscilloscope. For best results a properly terminated 75S cable should be used.

6.5.4. XLR output (5V 110S)

This conforms electrically and mechanically with the AES3 [1] or balanced IEC958 standards.

6.6. Modulator External Input

Impedance	10kS.
Maximum level	2.00V pk-pk (This gives calibrated modulation levels)

6.7. Synchronization Output

The output sync is a DC coupled positive going square wave intended for triggering an oscilloscope. It is not suitable for use with SDIF-2 word clock inputs.

Impedance	75S		
Level	0.4V (approx)		
Frequency			
	`Frame mode'	FS	
	`Bit mode'	FS * 128	(6.144MHz at 48kHz sample rate)

6.8. Power Supply Inlet

The unit is mains powered requiring 90 to 260AC at 50-60Hz. The power on/off switch, IEC320 inlet connector and fuse holder are located on the rear of the unit.

The unit consumes less than 0.3A.

The fuse rating is 1AT, also marked on the rear panel.

6.9. Physical

The enclosure is 1U high and fits standard 19-inch racking. The unit is 250mm deep from the front panel surface, excluding connectors, and the front panel controls protrude no more than 25mm.

7. ALIGNMENT INSTRUCTIONS

The equipment should not normally require re-alignment. However if the unit has been repaired or damaged in some way it may need adjustment. The following adjustments should only be carried out by experienced personnel. During the warranty period these adjustments need to be performed by an agent of Prism Media Products otherwise the warranty is void.

All the following operations should be carried out with the unit locked to a digital audio interface signal of sample rate between 44kHz and 48.1kHz.

7.1. External calibration indicator

Feed a 1kHz tone of amplitude 2V pk-pk (707mV rms) to the external modulation input.
Adjust the potentiometer R12 until the two indicators are both illuminated.

7.2. External modulation source gain

Set the modulation source to external.
Feed a 1kHz tone of amplitude 2.00V pk-pk (707mV rms) to the external modulation input.
Set the front panel amplitude knob to 25.
While measuring the voltage at TP3 (with respect to GND) adjust the potentiometer R48 to make the voltage the same as the input voltage (2.00V pk-pk).

7.3. Internal modulation source level

Set the modulation source to SQR.
Set the front panel amplitude knob to 25.
While examining the voltage at TP3 (with respect to GND) adjust the potentiometer R50 until the peak to peak variation equals 2.0V.

7.4. Fine (x1ns) range adjustment

Set the modulation source to SQR (8kHz square wave).
Set the front panel amplitude knob to 0.
Select the x1ns range (ie. the amplitude button indicator should not be illuminated).
Select bit sync output (ie. the sync button indicator should be illuminated).
Either connect an oscilloscope to trigger from the sync output and examine the signal on the BNC output or use the Prism DSA-1 to measure the FS jitter at the output.
Set the front panel amplitude knob to 25.
Adjust the potentiometer R87 until the output jitter measures 25ns.

7.5. Coarse (x10ns) range adjustment

Set the modulation source to SQR (8kHz square wave).
Set the front panel amplitude knob to 0.
Select the x10ns range (ie. the amplitude button indicator should be illuminated).
Select frame sync output (ie. the sync button indicator should not be illuminated).
Connect an oscilloscope to trigger from the sync output and examine the signal on the BNC output. Adjust the scope delay so that one of the broad pulses of the preamble (approx 480ns long) is visible in the centre of the screen.
Set the front panel amplitude knob to 25.
Adjust the potentiometer R86 until 250ns of jitter is seen on the oscilloscope.

8. REFERENCES

[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)

[2] IEC-958 - `Digital Audio Interface' First Edition, International Electrotechnical Commission, Geneva, Switzerland (1989 March)

[3] EBU Tech. 3250 - `Specification of the Digital Audio Interface - 2nd Edition' EBU Technical Centre, Geneva, Switzerland (1992)

[4] EIAJ CP-340 - `Digital Audio Interface', Standards of Electronic Industries Association of Japan, 1987

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[7] Julian Dunn - `Jitter and Digital Audio Performance Measurements' pp 148-166 of `Managing the Bit Budget' the proceedings if the AES UK Conference, 16-17 May 1994.

[8] Dunn J., McKibben B., Taylor R. & Travis C. - 'Towards Common Specifications for Digital Audio Interface Jitter' Preprint 3705, presented at the 95th AES Convention, New York, October 1993

[9] Draft AES-3ID-xxxx - `AES Information Document for Digital Audio Engineering - Transmission of AES3 formatted data by unbalanced coaxial cable' *J. Audio Eng. Soc.*, Vol. 42 No. 7/8, insert (July/August 1994)

APPENDICES

APPENDIX 1 JITTER AND THE DIGITAL AUDIO INTERFACE.

The JM-1 is designed to operate on digital audio signals conforming with the professional two-channel digital audio interface format, normally know as the AES/EBU or AES3 [1] interface, and with the similar consumer format known often as SPDIF described in IEC958 [2]. (These, or very similar formats, are also described in other standards documents [3,4].) It is assumed that users of this instrument will be familiar with these formats - which will be referred to in this manual as the digital audio interface.

Jitter is the variation in the timing of a periodic event - such as a signal transition - from an ideal timing that the event would have if were perfectly regular. For example, a perfect jitter-free square wave has an exactly constant time delay between transitions. In practice each transition of a real square wave, with exactly the same mean frequency will occur slightly before or after the ideal. This variation is called jitter.

Jitter on the digital audio interface, i.e. interface jitter, can be linked to many problems related to a digital audio installation and has two distinct effects. If interface jitter builds up to exceed the tolerance of following equipment the interface will fail as a communications channel and data errors or loss of synchronization will occur. The second effect is for a degradation in the audio quality as a consequence of jitter at a digital to analogue converter (DAC), an analogue to digital converter (ADC), or a sample rate converter (SRC). This is called sampling jitter.

While interface jitter should be tolerated up to a relatively high level before data integrity is threatened, jitter of these levels at the point of sampling in ADCs and DACs (sampling jitter) will produce objectionable audible effects. There is a relationship between interface jitter and sampling jitter because sampling clocks are often derived from an interface signal. This is the normal mode of operation for most equipment, since each installation generally has only one master sync generator.

The AES3 receiving circuit normally contains a phase-locked loop (PLL) whose job is to track the incoming transitions and extract the data. Ideally this receiver PLL will have a wide bandwidth, and the resultant ability to track incoming jitter will provide tolerance to high levels of interface jitter. Unfortunately some professional equipment with ADCs and DAC also use this PLL to generate the sampling clock directly, rather than using a second PLL to provide a more stable clock suitable for that purpose. Interface jitter can then be transferred to the converter device either directly or with minimal attenuation.

The normal recommended tests on digital audio equipment [5] do not assess the effects of incoming jitter but methods have been developed that characterise equipment in this respect. These are described in the literature [6,7].

At the time of writing the digital audio interface specifications do not define jitter behaviour. Proposals have been made toward remedying this [8]. The proposals are for new specifications for intrinsic jitter, jitter gain, jitter transfer function and jitter tolerance. The last four parameters require a known source of jitter, such as the JM-1, for their measurement. Use of the JM-1 for these measurements is illustrated in section 5.

APPENDIX 2 MODULATOR OPERATION.

A clock, at 128 times the sampling frequency (128FS), is recovered from the digital audio interface signal applied to the input. This clock then has a variable delay applied to it and is used to re-time the transitions of the input data stream. The nominal period of this clock is referred to as one unit interval (UI) and so the maximum jitter that can be generated in this way is therefore one UI.

The fine (x1ns) range uses this technique but the coarse (x10ns) range requires more than 1UI of jitter. This is achieved by cascading of several stages so that 500ns - more than 3 UI - of jitter can be generated.

The recovered clock has some jitter before any modulation has been applied. Low frequency jitter on the signal applied to the input will be passed through to the output largely unattenuated as the jitter transfer function of the clock recovery circuit only begins to attenuate jitter above 20kHz. If a very low jitter signal is applied to the unit and the modulator control is set to zero then the remaining jitter is the intrinsic jitter of the JM-1.