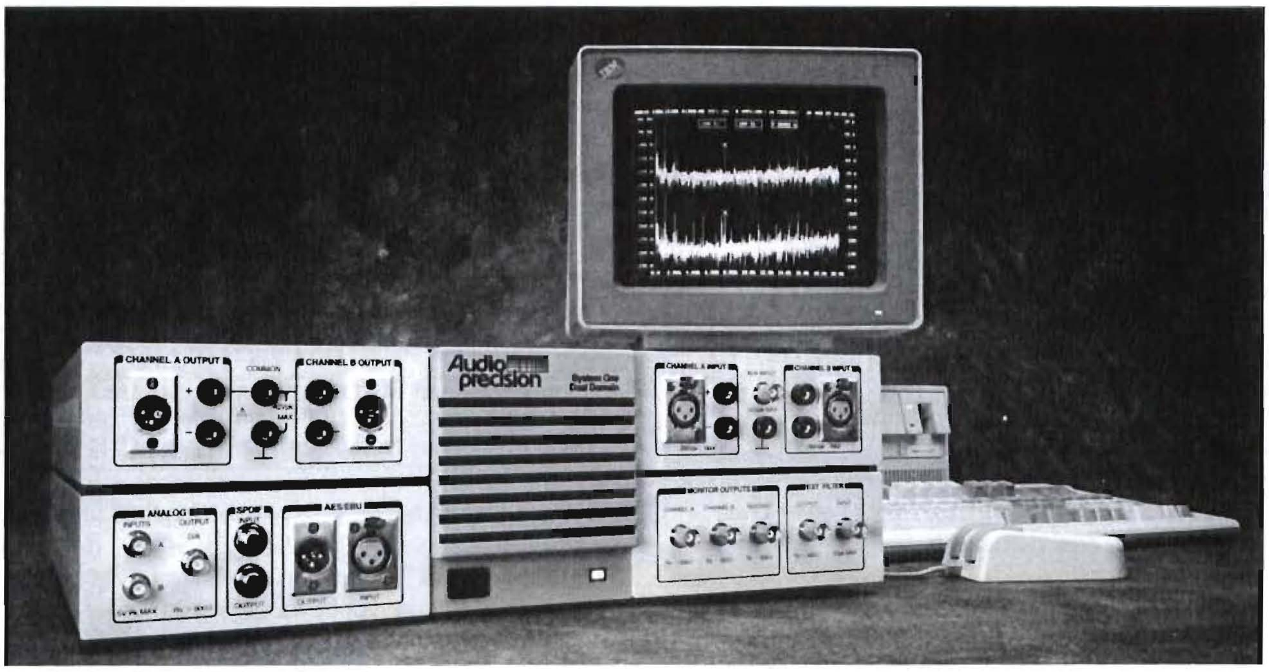
Audio precision

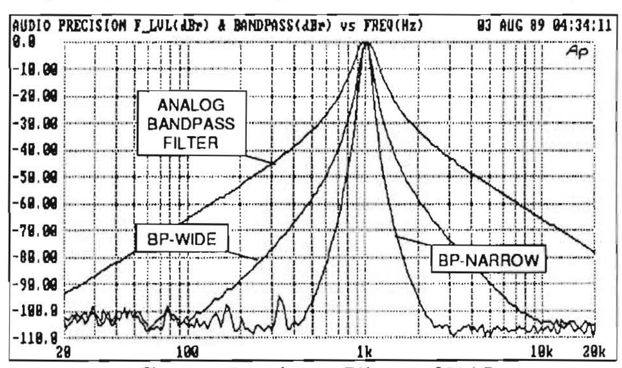
SYSTEM ONE + DSP
SYSTEM ONE DUAL DOMAIN



Audio Precision's new **System One + DSP™** (Digital Signal Processor) and **System One Dual Domain™** are revolutionary additions to the company's audio testing product line. They provide digitally-based analysis and synthesis of audio frequency signals. A reserved compartment in all System One mainframes permits these capabilities to be fitted into a System One already in service. **System One + DSP and System One Dual Domain** add significant capability to the already-impressive measurement features. **System One + DSP** expands previous applications with greatly enhanced measurements of analog signals. **System One Dual Domain** has these same capabilities plus, for the first time in any test instrument, the ability to generate and measure audio signals in any combination of digital and analog domains.

Enhanced Analog Measurement Capability

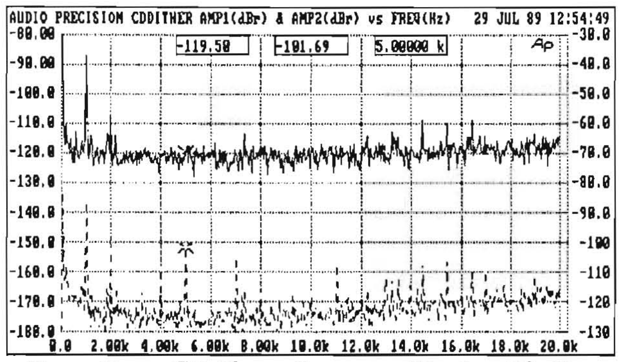
System One + DSP adds harmonic and spectrum analysis capability for users making analog measurements. By operating as a high-selectivity bandpass filter with two degrees of selectivity (see figure below) it may be used to measure individual harmonic distortion, depth of erasure of analog tapes, hum, and other wave analyzer-type measurements. The filter frequency steering source may be an operator-entered frequency, the analog generator frequency, or the analog BP/BR filter frequency. The filter may follow that source directly,



Response Curves, Bandpass Filters of HARMONIC.DSP (48 kHz Sample Rate) and Analog Analyzer at 1 kHz.

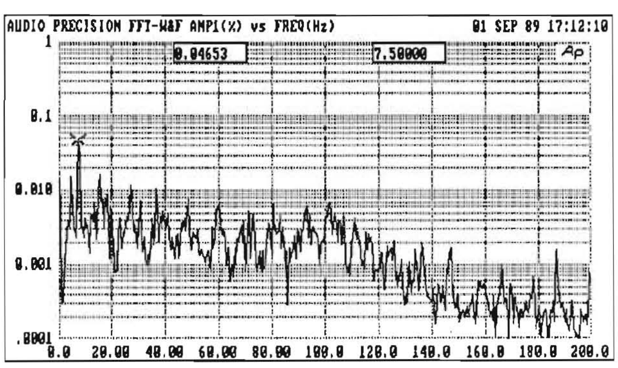
tune to the 2nd through 9th harmonic (selectable) of that source, or follow that source with any desired value of frequency offset.

System One + DSP may also operate as a dual channel waveform-digitizer/FFT-spectrum analyzer. The figure below shows the distortion and noise spectrum of a Compact Disc player reproducing the dithered -90 dB track (above) and undithered -90 dB track (below) of the CBS test disc. The reduced distortion and increased noise due to dither are readily seen and quantitatively measured with the new graphic cursor feature.



Upper Trace CD Player Spectrum, Playing -90 dB Dithered Test Track, CBS Test Disc. Lower Trace, -90 dB Undithered Test Track.

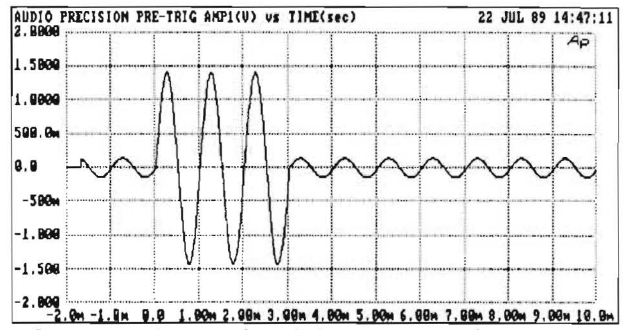
Signals up to a maximum frequency of 80 kHz may be acquired with 16 bit resolution and record length up to 30,720 samples per channel with the **MEM** option or **System One Dual Domain** or 8k samples per channel with standard memory in **System One + DSP**. Frequency domain displays are obtained by FFTs of up to 2k lines (8k lines with the **MEM** option or **System One Dual Domain**). The 8k lines will produce, for example, spectral analysis with approximately 3 Hz resolution across a 20 kHz bandwidth. For spectral



Spectral Analysis of Wow & Flutter, Professional Tape Recorder

analysis of wow and flutter (see figure below), approximately 0.06 Hz resolution is available across the bandwidth below 200 Hz.

Acquired waveforms may also be displayed in amplitude vs time (oscilloscope) format. One of the two FFT programs offers pre-trigger capability to retain a selectable amount of signal occurring prior to a trigger. See the figure below, acquired from System One's BUR option, for an illustration of pre-trigger and waveform display.

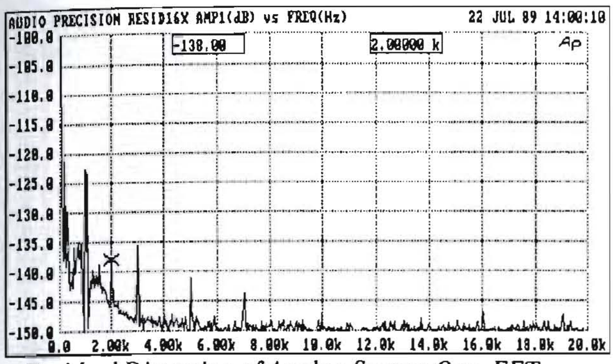


Sine Burst Acquired With 1.5 Milliseconds Pre-Trigger

Analog signals to be analyzed may be acquired from three sources:

- directly from dc-coupled fixed-range unbalanced input connectors on the DSP module front panel
- after the balanced inputs and gain range selection circuitry of the analog input section of System One
- from the READING MONITOR OUTPUT of System One (following all analog processing and filtering).

Signals acquired after the balanced inputs profit from the noise rejection of System One's high-CMRR inputs and have no high-frequency band limiting, but are not dc-coupled. Signals acquired after all analog processing can lead to extremely wide dynamic range measurements in many modes. For example, in THD+N mode System One's analog notch filter removes the fundamental sinewave signal, leaving only distortion and noise products. The DSP then converts this signal to the digital domain and further analyzes it via sharp bandpass filter or FFT, enabling distortion products to be resolved as low as 145-150 dB below the fundamental. The figure below shows the 1 kHz residual distor-



Residual Distortion of Analog System One, FFT With 16X Averaging

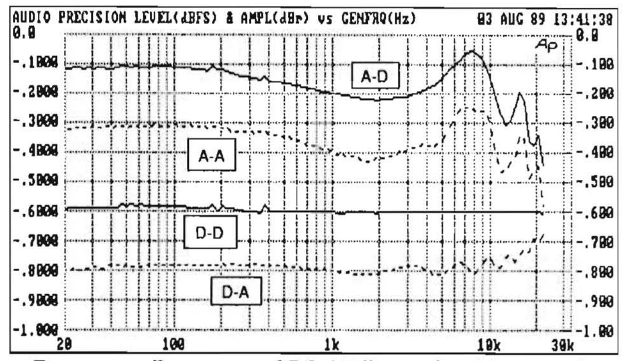
tion and noise of System One's analog generator and analyzer, plus the attenuated 1 kHz fundamental. This graph was obtained via selectable averaging of 16 acquisitions and FFT transforms.

Future DSP programs are planned for fast frequency response testing and acoustical measurements with **System One + DSP**.

Digital Domain Measurements

System One Dual Domain is the world's first test instrument to offer audio signal generation and analysis in both analog and digital domains. The world of digital audio is expanding at a rapid rate with new products introduced almost daily. A major stumbling block to their development and optimization has been the lack of test equipment which operates in the digital domain. **System One Dual Domain**, in addition to all the features of **System One + DSP** described above, has digital inputs and outputs which allow interfacing to most professional and consumer digital audio devices. Parallel and serial digital interfaces are provided. The parallel interface allows connection to many A-D and D-A converters and most DSP devices. The serial input and output support the complete specification of the AES/EBU digital interface and the Sony Philips Digital Interface (SPDIF) for consumer products. This allows testing professional and consumer digital audio equipment in the digital domain, separating the effects of the A-D and D-A conversion portions of the equipment under test.

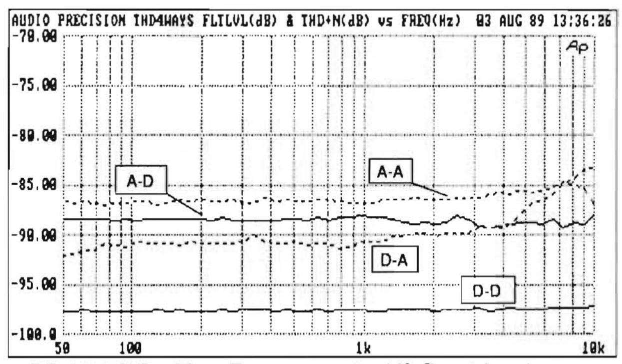
The figure below shows four overlaid frequency response measurements for the four possible combinations of analog and digital input and output of an RDAT recorder. The D-A curve shows ripple due to the oversampling digital filter in the RDAT output. The A-D curve shows response variations due to analog



Frequency Response of RDAT Recorder, Measured in All Four Domain Combinations

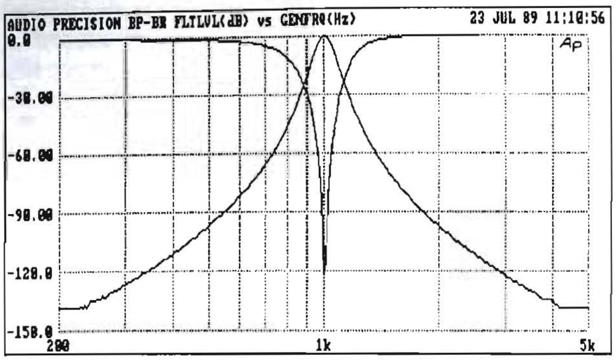
input filtering and the A-D converter. The A-A curve shows input and output filter ripple added.

The figure below shows THD+N versus frequency measurements of an RDAT for the four possible input-output combinations. The A-D and A-A curves show the higher distortion due to the A-D converter. The D-D curve is essentially at the -98 dB theoretical value for a 16-bit PCM digital audio system.

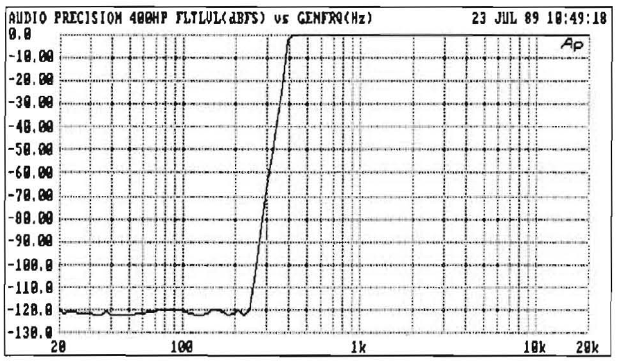


RDAT THD+N vs Frequency In All Combinations of Analog and Digital Inputs and Outputs

Signal processing in the digital domain can follow the same techniques as analog signal processing, except digitally implemented. Digital processing techniques available include RMS and quasi-peak detection, high pass filters, A-weighting and CCIR weighting filters for noise measurements, a bandpass filter for selective amplitude measurements, and a notch filter for THD+N measurements. Digitally-implemented bandpass, notch, and 400 Hz high pass filters are shown in the figures below. Digital-domain signals may thus be analyzed by techniques directly comparable to conventional analog signal analysis.



Bandpass and Bandreject Filter Resonse, GENANLR.DSP

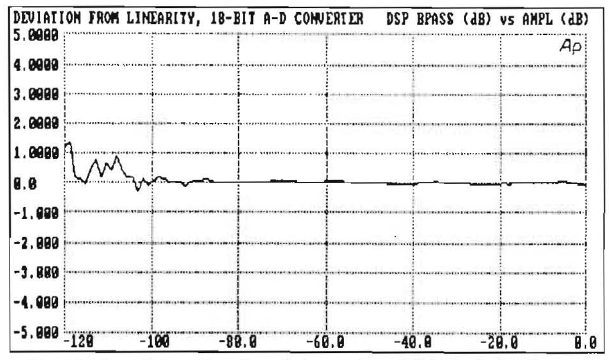


Response of 400 Hz Filter Selection

Data Manipulation and Presentation

DSP-processed measurement results may be graphed, compared to limits, and otherwise manipulated with standard System One software in the same manner that analog measurements are handled. The graph below shows deviation from perfect linearity of an 18-bit A-D converter. System One's (analog) generator drove the A-D converter input in a fixed-frequency 120-dB amplitude sweep. The digital output signal was acquired through the parallel digital interface, digitally bandpass filtered, and its amplitude digitally measured in the DSP module. The amplitude was plotted versus generator amplitude by System One software in the PC, and the COMPUTE LINEARITY function used to calculate and graph deviation from perfect linearity. The digital stimulus available from the digital outputs of **System One Dual Domain** may be swept in an identical manner to System One's analog generator. DSP-processed results may be saved to disk.

Screen-displayed signals may be precisely analyzed with help from the new graphics cursor feature of Sys-



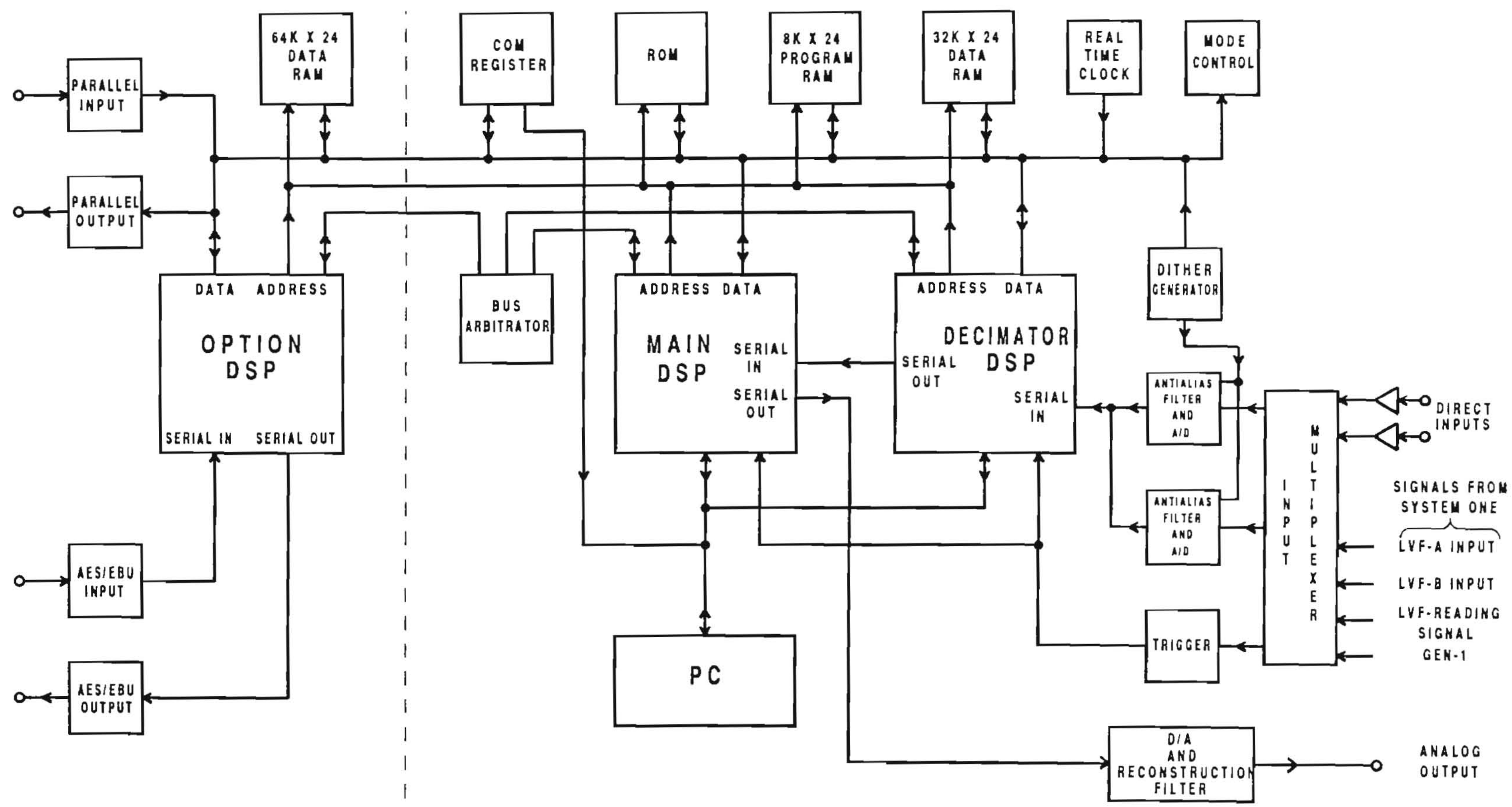
Deviation from Perfect Linearity, 18-Bit A-D Converter

tem One's new software version. Touching a horizontal arrow key (or mouse) brings a cursor and two or three numeric display windows onto the screen. The windows show the horizontal axis value and the vertical intercepts of the cursor with the one or two data lines displayed on the graph.

Digital signal processing programs are downloaded from PC disk to the DSP chip and DSP program memory in **System One + DSP** and **System One Dual Domain.** This permits many more types of DSP programs to be available than could be stored in the chip, and enables future programs to be easily added.

System One + DSP provides analog input and output capability and data memory for 8k samples. **System One Dual Domain** or the **MEM** option expands that memory for 30,720 samples. **System One Dual Domain** also provides both serial and parallel digital inputs and outputs.

Retrofits of DSP capability into System Ones already in use must be performed at the Audio Precision factory or at an authorized Audio Precision International Distributor.



DIGITAL I/O OPTION

MAIN BOARD

System One + DSP™ and System One Dual Domain™ Basic Specifications

PROCESSING Two 24 bit 25 MHz third generation digital signal processors

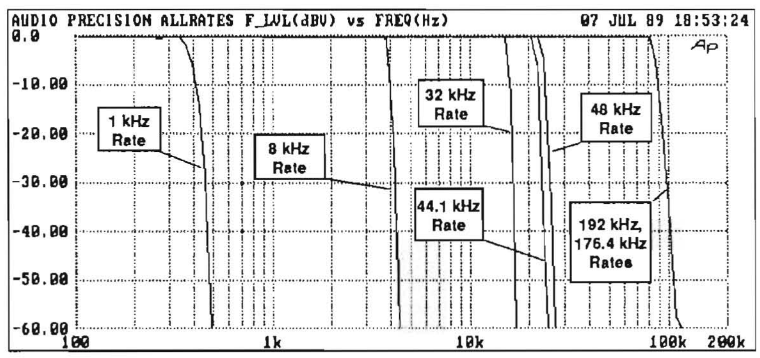
DATA MEMORY 32k x 24 bit (128k x 24 bit in **System One Dual Domain©** or the **MEM** option). Actual data record length depends upon DSP program in use.

PROGRAM MEMORY 8k x 24 bit

ANALOG INPUT RELATED

Converters: Dual channel independent 16 bit

Sample rates: 192k (80 kHz analog bandwidth), 176.4 k (80 kHz bandwidth), 48k (22 kHz bandwidth), 44.1 k (20 kHz bandwidth), 32 k (15 kHz bandwidth), or 1k sample/second (350 Hz bandwidth). See the figure below for typical frequency response at each sample rate. Not all sample rates are available with all DSP programs.



Analog source: Selectable A-monitor output, B-monitor output, Reading monitor output (analyzer output), generator monitor output, or front panel dc-coupled fixed-sensitivity inputs.

	Direct Inputs	Input via Analog Analyzer
Amplitude range	2.00 Vrms full scale (2.828 Vpk)	300 µV to 160 V rms, autoranging
Accuracy, flatness	±0.25 dB dc-(0.45 x sample rate) at sample rates ≥8 kHz; for example, dc-20 kHz @ 44.1 kHz sample rate	±0.25 dB 20 Hz-(0.45 x sample rate) at sample rates ≥8 kHz; for example, 20 Hz-20 kHz @ 44.1 kHz sample rate
Worst-case harmonic or spurious product	-90 dB for in-band signals (<0.5 x sample rate); -60 dB for out-of-band signals	

ANALOG OUTPUT RELATED

Converter: 16-bit, slaved to A/D sample rate.

Signal Routing: From front panel dc-coupled output, or through analog generator transformer-coupled output stage

System One Dual Domain™

System One Dual Domain™ has all the capabilities of **System One + DSP™** described above, and adds a third 24 bit 25 MHz digital signal processor, parallel and serial digital I/O, and expanded data memory to 128k words.

Parallel I/O

24 bit dual channel available on two 34 conductor connectors on rear panel (one for input, one for output). Channels are multiplexed on each connector. Data rates are selectable 32k, 44.1k, or 48k. Data strobe is included or may be externally supplied.

Serial I/O

Supports the full implementation of the AES/EBU digital interface. 20/24-bit data, parity, validity, and channel status bits are provided. The user bits are not supported. Electrically compatible with the Sony Philips Digital Interface (SPDIF). The transmitter and receiver may operate at 32k, 44.1k, or 48k. The transmitter may be slaved to the received signal, internal clocks, or house synch.

Functional Description and Specifications

The function of the DSP modules depends upon which DSP program is downloaded from the computer disk to the DSP unit. Initially, four DSP programs are available to perform individual harmonic analysis and other selective "wave analyzer" functions (HARMONIC.DSP), waveform display (time domain) and spectrum analysis (frequency domain) via fast Fourier transform (FFTGEN.DSP and FFTSLIDE.DSP), and generation and analysis of signals directly in the digital domain (GENANLR.DSP).

HARMONIC.DSP is a real-time program providing frequency-selective amplitude measurements of analog signals. A tunable bandpass filter may be steered by a panel entry, by the analog generator frequency, or by the analog analyzer bandpass-bandreject filter frequency. Tuning can be directly at the steering source frequency, at the 2nd through 9th harmonic of that frequency, or offset by a user-entered value above or below that frequency. A DSP-implemented RMS detector follows the filter.

FFTGEN.DSP and **FFTSLIDE.DSP** are programs for acquisition of waveforms and either waveform display or fast Fourier transform and spectral display. FFTGEN includes a digital sinewave generator function, can average FFTs for noise reduction purposes, and has modest triggering capability when acquiring signals. FFTSLIDE features more powerful and flexible triggering including pre-trigger, plus the ability to perform an FFT starting at any selected point in the stored signal. Both permit high-resolution FFTs with up to 8,192 spectral lines (bins), providing resolution of about 3 Hz at the 48 kHz rate and 0.06 Hz at the 1 kHz rate for wow and flutter analysis.

GENANLR.DSP is a real-time digital input-output program designed for use only with System One Dual Domain. It acquires digital-format audio data on two channels simultaneously, has two DSP-implemented RMS detectors, offers a selection of filters including tunable bandpass, tunable bandreject, and A-weighting and CCIR weighting filters plus a quasi-peak detector for noise measurements. It thus emulates in the digital domain most of the common analog domain audio measurements.

Digital Sinewave Generator Specifications (FFTGEN.DSP & GENANLR.DSP)

	Direct D-A Output	Through Analog Generator Output Stage	Digital Output
FREQUENCY RELATED			
Range vs Sample Rate	10 Hz-15 kHz @ 32 kHz; 10 Hz-20.67 kHz @ 44.1 kHz; 10 Hz-22.5 kHz @ 48 kHz		
Resolution	(Sample Rate)/2^{24} (approx. 0.003 Hz @ 48 kHz rate)		
AMPLITUDE RELATED			
Range	Fixed, 2.00 V_{rms} maximum	26.66 Vrms to <25 μV rms (same as analog generator)	Full 24 bit amplitude range
Resolution	$1/2^{16}$ (approx 30 μV)	<0.01 dB or 1.27 μV, whichever is greater	1 LSB ($1/2^{24}$)
Flatness and Accuracy	±0.25 dB 20 Hz-20 kHz @ sample rate ≥ 44.1 kHz		unmeasurable
Units	%FS, dBFS (FS=digital full scale)	same as analog generator	%FS, dBFS (FS=digital full scale)
DISTORTION			
THD+N	≤0.01%		≤0.00003%FS
Worst harmonic/spurious signal	-90 dB for in-band signals (<0.5 x sample rate); -60 dB for out-of-band signals		≤0.00003%FS
OUTPUT RELATED			
Configuration	Unbalanced, single channel	BAL-UNBAL-CMTST; FLOAT-GROUND; A, B, A&B, A&-B (same as analog generator)	AES-EBU, SPDIF, or parallel; A, B, A&B
Source Impedance	560 Ω	50/150/600 Ω (same as analog generator)	110Ω AES/EBU 75Ω SPDIF 22Ω parallel
DITHER			
Distribution	Triangular probability distribution		
Amplitude	± 16th bit		Selectable from ±1 LSB of 8 bit word through 24 bit word or OFF

Spectrum and Waveform Display (FFTGEN.DSP & FFTSLIDE.DSP)

	FFTSLIDE.DSP	FFTGEN.DSP
RECORD LENGTH		
Standard memory	8,192 samples/channel	4,096 samples/channel maximum
Duration @ 192 kHz	0.043 sec	0.021 sec
Duration @ 48 kHz	0.171 sec	0.085 sec
Duration @ 32 kHz	0.256 sec	0.128 sec
Duration @ 8 kHz	1.024 sec	0.512 sec
Duration @ 1 kHz	6.144 sec	4.096 sec
Maximum memory (MEM option or System One Dual Domain)	30,720 samples/channel	16,384 samples/channel maximum
Duration @ 192 kHz	0.160 sec	0.085 sec
Duration @ 48 kHz	0.640 sec	0.341 sec
Duration @ 32 kHz	0.960 sec	0.512 sec
Duration @ 8 kHz	3.840 sec	2.048 sec
Duration @ 1 kHz	24.576 sec	16.384 sec
TRIGGER RELATED		
Source	All analog or digital input sources, analog generator sync, power line	Channel 2 signal, which may be any analog or digital input source
Slope	+ or -	+
Pre-Trigger	Yes	No
FREQUENCY RESOLUTION	(Sample rate)/(FFT input data length); for example, 48 kHz sample rate and 16,384 samples gives 2.93 Hz resolution (bin width)	
SPECTRUM AVERAGING	No	1, 4, or 16x

AMPLITUDE RELATED	**Direct Inputs**	**Input via Analog Analyzer**
Accuracy, Flatness	Depends upon frequency separation of signal component from center of bin. Worst-case errors are approximately 0.8 dB for BH4 window, 1.5 dB for Hann, and 4.5 dB for "flat" (no window).	
Units (%FS & dBFS also available for analysis of digital signals in Dual Domain units)	V, dBV, dBr, dBm, dBu, W	V, dBV, dBr, dBm, dBu, W, %, dB, PPM, X/Y
CHANNEL PHASE MATCH	≤1° to 50 kHz	≤2° to 50 kHz
SPEED		
Signal Acquisition Time	see RECORD LENGTH/Duration above	
Transform, Windowing, and Magnitude Calculation	typically 600 msec for 16,384 samples;165 msec for 4,096 samples;50 msec for 1,024 samples	

Spectrum and Waveform Display (FFTGEN.DSP & FFTSLIDE.DSP)

Transfer to computer and display	depends on number of points plotted, computer processor type, clock rate, co-processor, type of display system. For 20 Mhz 80386 with 80387 co-processor, color VGA, 512 points plotted, typical time is 2.2 seconds

Harmonic Analysis (HARMONIC.DSP)

	Direct Input	Input via Analog Analyzer
AMPLITUDE RELATED		
Units	V, dBV, dBr, dBm, dBu, W	V, dBV, dBr, dBm, dBu, W, %, dB, PPM, X/Y
FREQUENCY RELATED	**Sample Rate ≤48 kHz**	**Sample Rate ≥176.4 kHz**
Range	10 Hz-21.77 kHz @ 48 kHz; 10 Hz-20.0 kHz @ 44.1 kHz 10 Hz-14.5 kHz @ 32 kHz	10 Hz-80 kHz
Filter Shapes	1/8 octave (Q=12, -3 dB BW 8% of center frequency) or 1/10 octave (Q=15, -3 dB BW 6.7% of center frequency)	1/8 octave (Q=12, -3 dB BW 8% of center frequency)
Filter Steering	Steering source software panel entry or analog generator frequency or analog analyzer BP/BR filter frequency. Filter can track directly at source frequency, or at selectable harmonic 2-9 of source frequency, or at panel-entered frequency offset above or below source frequency	

Digital Analyzer Specifications (GENANLR.DSP)

INPUT FORMATS	AES-EBU, SPDIF, Parallel. 2 channels, 24 bits
LEVEL MEASUREMENT RELATED	
Range	0 dBFS to -125 dBFS
Resolution	±0.01 dB
Accuracy & Flatness	±0.02 dB
Units	%FS, dBFS
FILTERED LEVEL MEASUREMENT RELATED	
Filter Shapes	Bandpass (Q=12), bandreject, BR+400 Hz HP, 400 Hz HP, A weighting, CCIR weighting
Bandpass Frequency Range	0.04% to 40% of sample rate; for example, 20 Hz-19.2 kHz @ 48 kHz sample rate
Bandreject Frequency Range	0.1% to 40% of sample rate; for example, 50 Hz-19.2 kHz @ 48 kHz sample rate
Residual THD+N	-120 dB
Units	%FS, dBFS, BITS, dB (ref LEVEL measurement)
FREQUENCY MEASUREMENT RELATED	
Range	5 Hz to 40% of sample rate for rated accuracy
Resolution	Maximum of 0.003% of reading or 0.0001% of sample rate
Accuracy	0.01% of reading or 0.0001% at 4 readings/second

Audio Precision, Inc., P.O. Box 2209, Beaverton, OR 97075
(503) 627-0832 1-800-231-7350 FAX (503) 641-8906
Telex 283957 AUDIO UR

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