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DSP NEWSLETTER

Volume 2, Number 1
July 10, 1989

AN ANNIVERSARY

This issue is the first anniversary of the *DSP NEWSLETTER*. We hope that you have found it useful and informative. Your comments and suggestions are always welcome. We'll be busy this summer expanding our product line to include floating point DSP products, DSP support for Steve Job's NeXT computer, and more products for DSPnet™, our exclusive multi-master 32 bit parallel interface bus.

NOW, 25 MFLOP FLOATING POINT DSP

We'll be shipping the most advanced floating point DSP card later this summer -- the DSP-32C. It starts with AT&T's 25 MFLOP DSP32C 32 bit floating point DSP chip. Next we add generous amounts of zero wait-state RAM (64 or 256 Kbytes) and two professional-audio quality analog I/O channels. On top we sprinkle SCSI and 32 bit DSPnet digital interfaces. That's the recipe for solving the most demanding signal processing problems. For software support, you should know that the DSP-32C is compatible with AT&T's development tools, including an assembler/link editor, C compiler and simulator/debugger. We will also be adding further software and hardware support for the DSP-32C to the menu in the near future.

For many DSP applications, there is no substitute for floating point. Integer calculations must be carefully planned so that overall accuracy is not lost due to truncation or overflow. With floating point DSP, this is handled automatically. The DSP-32C provides 32 bit accuracy (24 bit mantissa, 8 bit exponent) and a peak performance of 25 MFLOPS. Also, unlike some Brand "X" floating point DSP's, the DSP-32C can convert to IEEE format floating point in a single 80 nanosecond cycle. This makes the DSP-32C much more efficient as a host coprocessor.

System performance is also enhanced by closely coupling the I/O resources to the DSP chip. The DSP-32C is a completely self-contained real-time data acquisition, signal processing, and signal generating system. Even while maintaining real-time analog I/O, the DSP-32C can perform DSP-to-host, DSP-to-SCSI and DSP-to-DSPnet transfers. Developing applications is also faster and easier with a complete DSP system on a single board. For multiprocessing applications, DSPnet is a fast and convenient means of interconnection. For large databases of signals, the built-in SCSI port provides high speed access to virtually unlimited mass storage.

We hate to keep reminding you that all 16 bit analog interfaces are not created equally. Many other systems out there charge you for 16 bits without delivering. Ariel's DSP-32C provides true 16 bit performance, with a signal-to-noise ratio in excess of 90 dB. Beware of systems that spec 80 to 85 dB dynamic range, for that's only 13 or 14 bits of precision! In addition to a low noise floor, check for brick wall, linear phase, tracking antialias filters, simultaneous sampling of both input channels, industry-standard sample rates, and electronically balanced inputs with high quality connectors. Need we say more?

BUG-56 PORTED TO NeXT COMPUTER

Unless you have been living under a rock, you have probably heard of Steve Job's latest opus: the NeXT workstation. One thing that differentiates this from the other UNIX-based workstations is that every NeXT is equipped with a 25 MHz DSP56001 DSP chip from Motorola. The NeXT cube also incorporates a microphone

Ariel Corporation ▪ 433 River Road ▪ Highland Park, NJ 08904 ▪ Telephone: 201-249-2900
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If you have any questions or you want to learn more about our products, tear off this page and return it to us in the reply envelope. Overseas customers, please attach the appropriate postage!

Make any corrections or additions necessary to the address label above.

Please send more information regarding:

[ ]	**DSP-32C**	AT&T's DSP32C based DSP and 16 bit analog I/O for IBM PC
[ ]	**DSP-56**	DSP56001-Based Data Acquisition Processor
[ ]	**DSP-16 Plus**	New, Lower Noise Level Data Acquisition Processor
[ ]	**FDB-20**	Filter Daughter Board for the DSP-16 Plus
[ ]	**SDI**	Signal to Disk Interface Recording System for IBM PC
[ ]	**SYSid**	Acoustic Testing System for IBM PC
[ ]	**FFT-320**	FFT Subroutines Package for DSP-16
[ ]	**DSP-300**	Digital Signal Coprocessor with FFT's for HP 200/300
[ ]	**ADC56000**	Analog I/O for Motorola DSP56000ADS Development System
[ ]	**DSP-C25**	TMS320C25 Low Cost Coprocessor Card for IBM PC
[ ]	**PC-56**	DSP56001 Low Cost Coprocessor Card for IBM PC
[ ]	**BUG-56**	Advanced Monitor/Debugger for Ariel's DSP56001 Cards
[ ]	**SA-1620**	Real Time FFT Spectrum Analyzer

[ ] **Other** ______________________________

[ ] **Please call** Phone: ____________________ Best Time: ____________

Comments: ____

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COMING SOON TO A SLOT NEAR YOU

- The NeXT step in Digital Audio
- Faster Integer DSP
- More Products for DSPnet

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Export Price List

DSP-32C

DSP-32C is a Floating Point DSP Development System with pro-audio quality analog input/output. It is based on AT&T's 32 bit DSP32C DSP chip. The DSP-32C is software and hardware compatible with AT&T's DSP32C-DS DSP Development System and DSP32C software development tools.

DSP-32C	Floating Point DSP Development System	$4795.00
64K-32C	Upgrade standard 16 Kword memory to 64 Kword	955.00
DSP32C-SL	Software Support Library (Assembler, Linker, etc.)	600.00
DSP32C-CC	Optimizing C Language Compiler	1800.00
DSP32-AL	DSP32 Family Application Library	115.00

DSP-16 PLUS

DSP-16 Plus is a complete TMS320 Development System with pro-audio quality analog input/output, and includes the DSP-16 Hardware and PDS-320 Program Development Software for the TMS32020 and TMS320C25.

DSP-16 Plus	Real-Time Data Acquisition Processor/Development System	$2995.00
C25-16	TMS320C25 (40 MHz) Processor Option for DSP-16	600.00
2MB-16	2 Megabyte (1 Megword) Data Buffer Option for DSP-16	1200.00
FDB-20	Filter Daughter Board (Third Party Installed)	899.00
DIO-16	Digital I/O Prototype Card for DSP-16	180.00

SDI

SDI Signal-to-Disk Interface Systems include the DSP-16 Hardware, C25-16 TMS320C25 Processor Option, a SCSI Controller, DSPDISK and DSPDOS Software, and mass storage devices and cabling as applicable.

SDI-0E	SDI for User Supplied Mass Storage Devices	$4195.00
SDI-80I	SDI with 80 Mbyte Internal SCSI Disk Drive	5995.00
SDI-80E	SDI with 80 Mbyte External SCSI Disk Drive	6475.00
SDI-300I	SDI with 300 Mbyte Internal SCSI Disk Drive	8995.00
SDI-300E	SDI with 300 Mbyte External SCSI Disk Drive	9475.00
EXT-80	Additional 80 Mbyte SCSI Disk Drive with Cables and Enclosure	2395.00
EXT-300	Additional 300 Mbyte SCSI Disk Drive with Cables and Enclosure	5395.00
SDI-L	Source Code License for DSPDISK and DSPDOS	600.00
FDB-20	Filter Daughter Board (Third Party Installed)	899.00
UPG-020	SDI Upgrade for DSP-16 with TMS32020 Processor	subtract $1995.00
UPG-C25	SDI Upgrade for DSP-16 with TMS320C25 Processor	subtract $2495.00

SYSID

SYSid Audio-Bandwidth Test Systems include the DSP-16 Hardware, C25-16 TMS320C25 Processor Option, and SYSid Software.

SYSid	Complete Audio-Bandwidth Test System	$3595.00
FDB-20	Filter Daughter Board (Third Party Installed)	899.00
UPG-16	SYSid Upgrade for DSP-16 with TMS32020 Processor	subtract $1995.00
UPG-16C	SYSid Upgrade for DSP-16 with TMS320C25 Processor	subtract $2495.00

SA-1620

SA-1620 Turnkey FFT Audio Spectrum Analyzer includes the DSP-16 Hardware and SA-1620 Software.

SA-1620	Turnkey FFT Audio Spectrum Analyzer	$2995.00
FDB-20	Filter Daughter Board (Third Party Installed)	899.00
UPG-SA	SA-1620 Upgrade for any DSP-16	subtract $1995.00

DSP-C25

DSP-C25 is a low-cost full speed DSP coprocessor based on the Texas Instruments' TMS320C25 DSP chip. DSP-C25 includes a program upload utility and a demo program. DSP-C25 includes 16 Kwords memory. For complete memory expansion order one 64K-C25 option and one 64U-C25 option.

DSP-C25	Signal Processing Card with 40 MHz TMS320C25 Processor	$715.00
AIO-14B	Optional 14-bit Analog I/O	115.00
16K-C25	Additional 16 Kword Data Memory	115.00
64K-C25	Additional 64 Kword Data Memory	595.00
64U-C25	Upgrade 16 Kword Program Memory to 64 Kwords	475.00

DEVELOPMENT TOOLS AND APPLICATIONS SOFTWARE FOR TMS320 PRODUCTS

ASM-320	Ariel's Macro Assembler for Texas Instruments' TMS32020/TMS320C25	$49.95
TI-C25	Texas Instruments' TMS320C25 Macro Assembler	600.00
PDS-320	Complete Program Development Software for DSP-16 and DSP-C25 (includes ASM-320)	600.00
FFT-320	FFT Subroutines for the DSP-16 and DSP-C25	360.00
FIDAS	Momentum's FDAS Filter Design Package with DSP-16 Drivers	835.00
HWS-320	Hyperception's Hypersignal-Workstation Software for the DSP-16	1389.00

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DSP-56

DSP-56 is a 24 bit integer DSP development system with pro-audio quality analog input/output. DSP-56 includes a program upload utility and a demo program, and is compatible with the BUG-56 symbolic debugger and other DSP development tools listed below. DSP-56 includes 16 Kwords memory. For complete memory expansion, order two 64K-56 options and one 64U-56 option.

DSP-56	24 Bit DSP Card with Motorola DSP56001 Processor and 16 Bit Analog I/O	$3595.00
16K-56	Additional 16 Kword memory (two per DSP-56 max.)	175.00
64K-56	Additional 64 Kword memory (two per DSP-56 max.)	895.00
64U-56	Upgrade standard 16 Kword memory to 64 Kwords (one per DSP-56 max.)	715.00

PC-56

PC-56 is a low-cost full speed DSP coprocessor using the Motorola DSP56001 DSP chip. PC-56 includes a program upload utility and a demo program, and is compatible with the BUG-56 symbolic debugger and other DSP56001 development tools listed below. PC-56 includes 16 Kwords of memory.

PC-56	Signal Processing Card with Motorola DSP56001 Processor	$715.00
AIO-14B	Optional 14-bit Analog I/O	115.00
64U-56	Upgrade standard 16 Kword memory to 64 Kwords	715.00

ADC56000

ADC56000 is a two channel 16 bit analog input/output subsystem for Motorola's DSP56000ADS DSP Development System. ADC56000 includes utility and demo software and schematics.

ADC56000	Analog I/O for the Motorola DSP56000ADS Development System	$1195.00
FLT-10K	Anti-Aliasing Filters (10.25 kHz) Option for ADC56000	300.00
FLT-20K	Anti-Aliasing Filters (20 kHz) Option for ADC56000	300.00

DEVELOPMENT TOOLS AND APPLICATION SOFTWARE FOR DSP56001 PRODUCTS

BUG-56	Ariel's DSP56001 Monitor/Debugger for PC-56	$475.00
DSP56000CLASA	Motorola DSP56001 Assembler/Simulator	595.00
DSP56KCCA	Motorola DSP56001 C Compiler	850.00

DSP-300 FOR HEWLETT-PACKARD SERIES 300

DSP-300 is a Motorola DSP56001-based DSP coprocessor for Hewlett-Packard Series 200/300 computers. DSP-300 includes a number of FFT and related functions, and BASIC and Pascal drivers with source code. HP-UX drivers are available optionally.

DSP-300	2nd Generation Signal Processing Card for HP 200/300	$2880.00
64K-300	64 Kword Memory Upgrade for DSP-300	895.00
UX-300	HP-UX Drivers for DSP-300	595.00

FFT-523 FOR HEWLETT-PACKARD SERIES 300

FFT-523 is a dedicated FFT coprocessor for Hewlett-Packard Series 200/300 computers. FFT-523 includes BASIC and Pascal drivers with source code.

FFT-523	Fast Fourier Transform Coprocessor for HP 200/300	$1920.00
	Hanning, Hanning Squared, Bartlett or Custom Window (please specify one)	240.00
	Initial Setup Charge for Custom Window	120.00

PC-FFT

PC-FFT is a dedicated FFT coprocessor for the IBM PC/XT/AT. PC-FFT includes BASIC driver with source code. C and Fortran drivers are available on request.

PC-FFT	Fast Fourier Transform Processor for the IBM PC	$2220.00
	Hanning, Hanning Squared and Bartlett Windows (combined), or Custom Window	240.00
	Initial Setup Charge for Custom Windows	120.00

OPERATING MANUALS

Product pricing includes one copy of the Operating Manual. Operating manuals for any of Ariel's products are available separately for $35.00. The cost of any manual may later be applied towards the purchase of that product.

These prices are applicable to all sales outside of the United States, Canada and Mexico. For US, Canada and Mexico, see North American Price List. Our prices are in U.S. dollars, FOB Factory, and include operating manuals and packing suitable for shipment by air freight. Shipping is via Air Parcel Post unless otherwise specified. Shipping charges are collect, or prepaid and added to invoice.

All Ariel manufactured equipment is warranted for one year from date of purchase from Ariel or an authorized dealer. Disk drives supplied as part of the SDI systems are warranted for 90 days from date of purchase from Ariel or an authorized dealer. The warranty covers parts, labor and return shipping.

May 1st, 1989 -- Pricing is subject to change without notice

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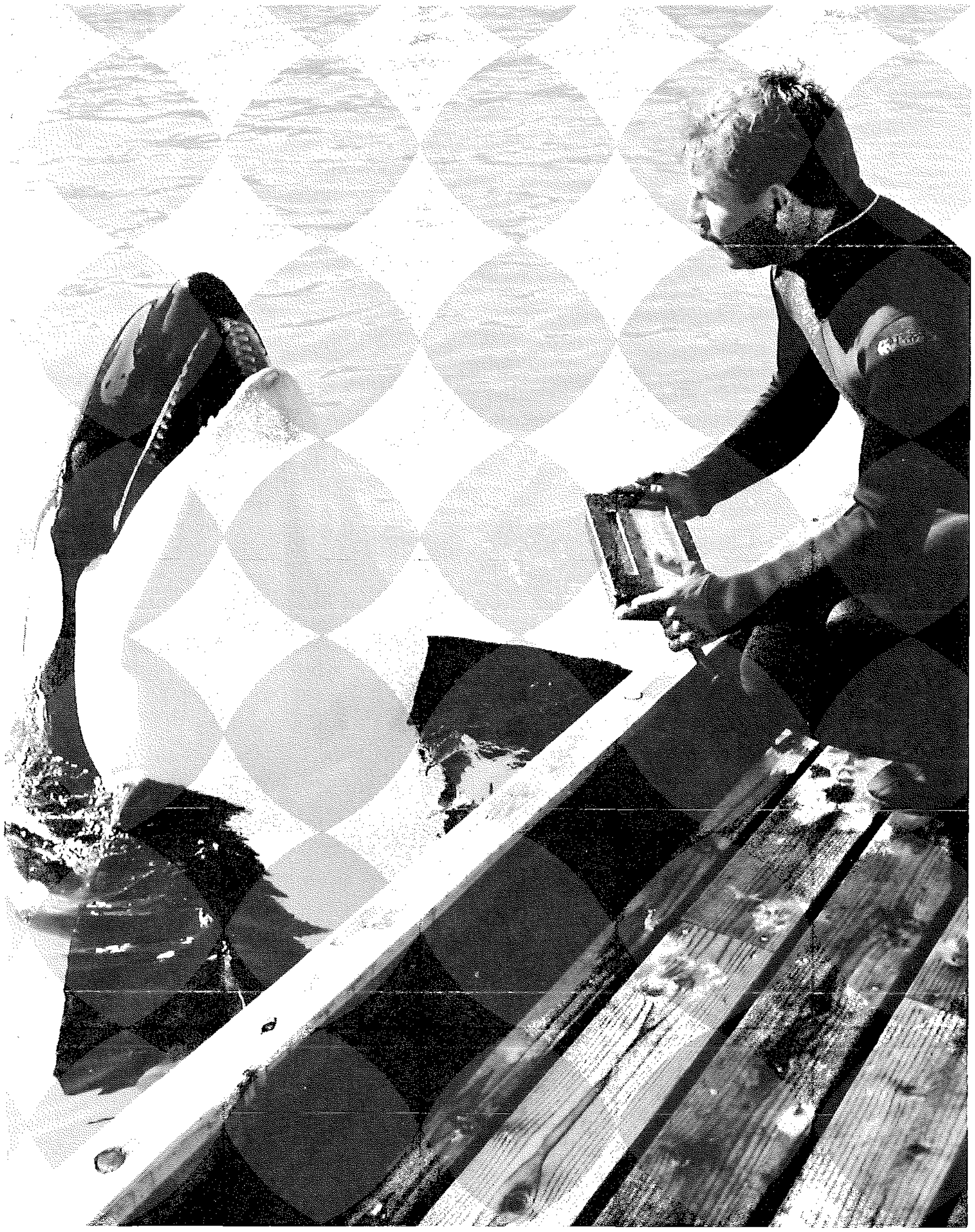
DSP NEWSLETTER

Volume 2, Number 2
Fall, 1989



A WHALE OF AN APPLICATION FOR THE ARIEL DSP-16

Picture courtesy of Sea World

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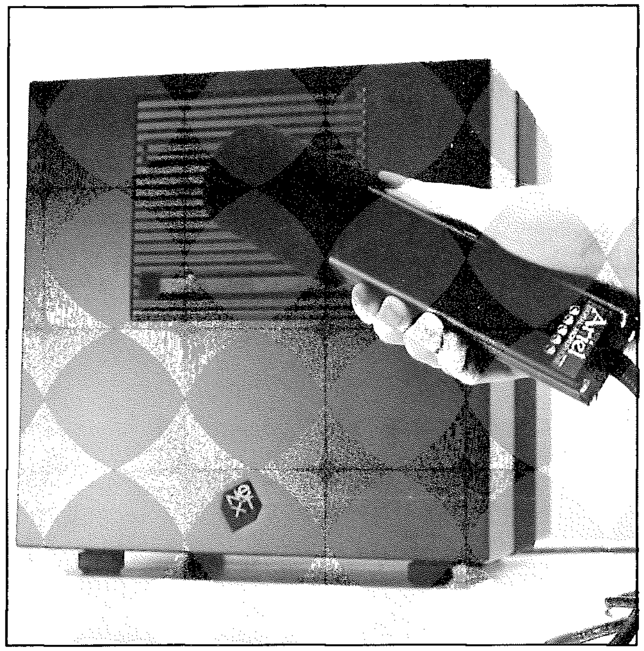
THE MICROPHONE COMES OF AGE

Ariel has introduced the world's first digital microphone, heralding the beginning of a new era in which the quality of input equipment will finally match the new capabilities of digital technology. The initial version of the Digital Microphone works exclusively with the NeXT™ Computer System, literally turning the system into a digital recording device for voice recognition, music and data acquisition applications.

Utilizing two 16-bit Motorola 56ADC analog to digital converters, the microphone's digital output is connected directly to the NeXT's DSP Port which utilizes the DSP56000. Benefits of this technique over the use of a conventional microphone include immunity from noise and interference generated from nearby electrical equipment and the ability to bandlimit the microphone through programmable sampling rates.

Applications are not limited to just voice recording. With dual input jacks, the microphone allows the NeXT to tap into the outside world. This unique feature allows the creation of multimedia applications by combining signals from voice, compact disk, or a mixing board. In addition, various transducers such as accelerometers and thermocouples may be connected to the dual inputs, transforming the NeXT into a precision data acquisition system.

While currently limited to the NeXT DSP Port, the microphone's digital advantages could easily be appreciated by the pro-audio industry as well. Who knows? Is there a AES/EBU-compatible microphone in the future? The curious should drop by our both at the AES show in New York, October 18-21.



THE ARIEL DIGITAL MICROPHONE

AT&T AND IBM MERGE

Following release of the floating point DSP-32 DSP board, Ariel is introducing the PC-32C, a low cost floating point Digital Signal Processing card for the IBM PC and compatibles. Utilizing AT&T's DSP32C floating point chip (25MFLOP) and a complement of zero wait-state RAM, it may be used alone as a high speed coprocessor/development system or as multiple boards linked via its serial I/O channel. Priced under $2000, the PC-32C is available with various memory options. For those of you who appreciated BUG-56, we will be offering a symbolic debugger for the DSP-32C and PC-32C: D3EMU.

FREE ! FREE ! FREE !

Yes, its true! Ariel will be giving away free 100% cotton, Pro-Quality, Polo shirts. The only requirement is that you submit a brief essay (200 words with pictures if possible) on your DSP application using Ariel's boards, systems or software packages. As space allows, these application articles will be printed in this newsletter. If fame and a prized Ariel polo shirt still doesn't entice you, we will give away a brand new Sony portable CD player to one lucky participant. A drawing will be held for all those who submitted articles two weeks prior to the publication of our Spring newsletter.

Get your articles in fast, first deadline for inclusion in our next newsletter, a free shirt and your chance at a free CD player is December 10, 1989.

ARIEL AND NeXT TIE THE KNOT

After a short but sweet romance, NeXT has decided to bundle Ariel's BUG-56 with its OS 1.0 operating system. BUG-56 is a menu-driven, window-oriented, symbolic debugger for use with the Motorola DSP56001. For those of you non-NeXT owners, you can add the *power* of the DSP56001 to your PC compatible platform with the Ariel DSP-56 or PC-56 DSP boards. Which by the way, are compatible with BUG-56. With all these neat applications being developed for the NeXT, wouldn't it be nice if a DSP Port was available for PC compatibles (who knows what the future holds)?

COMPLIMENTARY TICKETS (FREE FOR THE ASKING)

We have a limited number of free passes for the Audio Engineering Society's Convention and INFO'89 show in New York City. If you are interested in attending please give us a call so you can attend as our guests. If you are planning to attend, please drop by our both and say hello. We will be demonstrating our new Digital Microphone as well as many of our new products. In addition, we will have our engineering staff on hand to answer any of your technical questions.

THE NAME GAME

Product names can be quite confusing, especially around here. To better reflect product capabilities and families, we will be doing some fine-tuning and will keep you informed. As you are aware, for each of the major DSP chips we offer both a high level pro-quality full length development board and a low cost DSP coprocessor short card. To differentiate the two, we have standardized on calling them the "Model DSP-XXX" and the "Model PC-XXX" respectively. As an example, our product line supporting the AT&T DSP-32C consists of the Model DSP-32C our Pro-audio full size development board and our Model PC-32C low cost coprocessor short card. In light of this standardization we are renaming our DSP-C25 which is a low cost (high value) Texas Instrument's TMS320C25 based DSP card to the PC-C25. This full speed DSP coprocessor board complements the DSP-16 Plus, which is a development system with 16 bit analog I/O for the TMS320 DSP chip.

ARIEL'S DSP DEVELOPMENT BOARDS FOR THE PC PLATFORM

DSP CHIP	BITS	LOW COST SHORT CARD	PRO-AUDIO FULL FEATURED
Texas Instruments TMS320C25	16	* PC-C25	DSP-16 PLUS
Motorola DSP56001	24	PC-56	DSP-56
AT&T DSP32C	32	PC-32C	DSP-32C

** Formerly the DSP-C25*

■ ■ ■ CALENDAR ■ ■ ■

Oct 10 - 13 INFO, New York City

Oct 18 - 21 AES, New York City

Nov 2 - 5 ICMC, Columbus, Ohio

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WHAT ARE PEOPLE DOING WITH THIS STUFF, ANYHOW?

Applications for Ariel's products range from finding distressed aircraft via satellite monitoring to sound editing of local news broadcasts. We want to hear from you and share your applications with our readers. This application (submitted by David Reisner) involves the use of DSP in allowing humans to communicate with orcinus orca.

At the Sea World marine parks in California, Texas and Ohio, a system based on the Ariel DSP-16 is being used to train killer whales (orcinus orca). Taking advantage of the high quality analog outputs (16-bits) of the DSP-16, David Reiner Consulting, in association with the Sea World Research Institute, generates frequencies modulated by "syllables" and "words" from a "synthetic language" of up to 720 words. Cue words and sentences are composed on stage on a waterproof, hand held computer terminal. These underwater audio cues are used to replace or supplement the gestural cues which have been used in past training.

ARIEL'S STAFF IS GROWING

Bringing you the best in DSP Development Systems, Software and Applications is labor intensive. To continue to provide you with quality service and products, we are expanding. We would like to welcome two new engineers to our staff; Marc Lindahl and Steve Curtin. In addition, I would like to welcome myself, Les Listwa, who will serve as the editor of this newsletter and as Director of Marketing. If you have any questions or comments concerning this newsletter, or Ariel products please contact me directly or via our BBS.

If you have any questions or you want to learn more about our products, tear off this page and return it to us in the reply envelope. Overseas customers, please attach the appropriate postage!

Please make any corrections or additions to the address label above.

Please send more information on:

[ ]	**DSP-16 PLUS**	New, improved TMS320 based DSP and 16 bit analog I/O for IBM PC
[ ]	**DSP-32C**	AT&T's DSP-32C based DSP and pro-audio quality analog I/O
[ ]	**DSP-300**	Digital Signal Coprocessor with FFT's for HP 200/300
[ ]	**DSP-56**	DSP56001-Based Data Acquisition Processor for IBM PC
[ ]	**PC-56**	Low Cost DSP56001 Coprocessor Card for IBM PC
[ ]	**PC-C25**	Low Cost TMS320C25 Coprocessor Card for IBM PC
[ ]	**PC-32C**	Low Cost DSP-32C Coprocessor Card for IBM
[ ]	**FDB-20**	Brickwall filters for DSP-16 PLUS
[ ]	**BUG-56**	Advanced Monitor/Debugger for Ariel's DSP56001 Cards
[ ]	**D3EMU**	Advanced Monitor/Debugger for Ariel's DSP-32C Cards
[ ]	**PDS-320**	Program Development System for DSP-16 and DSP-C25
[ ]	**SYSid**	Acoustic Test System for IBM PC
[ ]	**SDI**	Signal to Disk Interface Recording System for IBM PC
[ ]	**SA-1620**	Real Time Spectrum Analyzer for IBM PC
[ ]	**FFT-320**	FFT Subroutines Package for DSP-16 and DSP-C25
[ ]	**ADC56000**	Analog I/O for Motorola DSP56000ADS Development System
	DIGITAL MICROPHONE	
[ ]	**DM-N**	All-Digital Microphone For NeXT Computer
[ ]	**OTHER**	I Would like To Interface The Digital Microphone To: ______________________

[ ] **Please call** Phone: ________________________ Best Time: ______________

Comments: ____

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DSP NEWSLETTER

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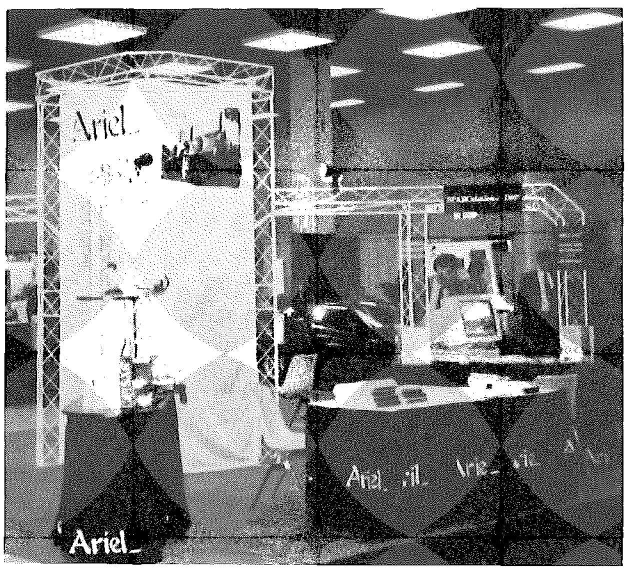
SPRING/SUMMER, 1990

GREETINGS AND SALUTATIONS!

We've recently merged our newsletter and customer databases, so if this is your first issue, welcome aboard. It's been a while since the last DSP NEWSLETTER, so we have much to report, including new DSP56001- and 96002-based DSP, digital audio interfaces, a speech spectrograph workstation, optical disk support for SDI, SPARC-based DSP, and even a major price reduction!



The Ariel Booth at ICASSP-90

A DESK-TOP SUPERCOMPUTER?

Well not quite, but how *do* you characterize a system that delivers 100 MFLOPs, 16 Mbytes of memory, and high-speed I/O on a single PC-based card? We call it the MM-96 and it's certainly not your typical DSP plug-in board. Based on a pair of Motorola's recently introduced 96002 DSP chips (*that's right they really do exist!*), the MM-96 transcends traditional DSP. In fact, it's far and away the most potent hardware Ariel has ever created.

The 96002 is the 32-bit floating-point big brother to Motorola's DSP56001 and builds on an architecture perfectly tailored for executing DSP algorithms (deterministic code execution, non-pipelined data ALU, fast interrupts). For DSP, it's power is unmatched: running at 33MHz it delivers 50 MFLOPs (165 MOPS!). It's also the first DSP chip that is 100% IEEE-compliant (IEEE 754 single precision and single extended precision).

But DSP is just the beginning! As a graphics/imaging transformation engine it compares favorably to devices like Intel's i860. And, recognizing that applications have an insatiable appetite for MFLOPs, Motorola has designed the part with multiprocessing in mind. The part sports *two* independent, symmetrical 32 bit ports each with its own DMA controller. Often in number crunching situations the processing engine is constantly interrupted by data movement, thus squandering precious FLOPs. The 96002's DMA capabilities provide a unique balance between processing and data transfer allowing data to move on and off the chip while the processor crunches continuously!

Given the unique competencies of the 96002, we were forced to re-think (and you know how painful that can be!) the concept of the *traditional* PC-based DSP card. The result? Ariel's MM-96. Dual 96002s, up to 16 Mbyte of memory, high speed I/O for audio (DSPnet™) and video (DT-*Connect*™) and a pair of expansion ports for *really* big problems. The expansion ports allow multiple MM-96s to be interconnected in a wide range of topologies; with each board contributing a full 100 MFLOP.

The MM-96 is ideal for multimedia, graphics, image processing, neural networks, simulation, audio processing, and algorithm development.

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THE POWER OF THE NeXT DSP PORT

The Ariel Digital Microphone with its dual-channel, pro-quality 16-bit analog-to-digital converters has created quite a stir, literally turning the NeXT computer into a digital recording device for voice, music, and data. Well, we've taken the *next* step (pun intended), bringing the digital microphone and the many capabilities of the NeXT DSP Port to the IBM PC.

The newly introduced PC-56D is a complete PC-based, low cost audio DSP workstation based on the 27 MHz, 13.5 MIPS Motorola DSP56001. The board contains not only a NeXT-compatible DSP port, which is completely compatible with our Digital Microphone and other NeXT DSP Port products, but also a 14-bit analog I/O chip with a microphone preamp. This combination is an ideal platform for low-cost audio DSP development.

MORE POWER FOR THE SUN

Developed for the SUN's SPARCstation1™, Ariel's two new SBus cards, the S-56 and S-56X, bring the power of DSP to the SBus. Based on the Motorola 27 MHz DSP56001, the Ariel SBus DSP cards are well suited to real-time processing tasks not normally possible in Unix-based environments. Both boards include NeXT-compatible DSP Ports for interface with external audio and data acquisition I/O. The S-56X includes a programmable Xilinx 3042 Gate Array chip for added functionality. The boards are ideal for use as dedicated integer array processors, real-time data acquisition and analysis systems and real-time controllers. The S-56 and S-56X will be available this summer. Device drivers and development software are in the works.

INTRODUCING SpeechStation™

Developed in conjunction with Sensimetrics Corporation, SpeechStation brings the power of a complete speech analysis system to the IBM-PC *at a very reasonable price*. SpeechStation is an easy-to-use mouse-driven, fully integrated package which combines Ariel's hardware with software created by the speech researchers at Sensimetrics. Among the many features of the system are real-time recording and playback, display and high quality printing of LPC and FFT spectrograms, waveform editing, pitch tracking, annotation, and comparison of selected speech segments. To find out more about the Sensimetrics SpeechStation, call or write for your free demo disk.

WHAT'S DAT BOARD?

The answer is the Ariel DAT-56, a complete high-quality digital audio signal processing system on a single IBM PC compatible card. Designed for interface with Digital Audio Tape (DAT) Recorders, CD players, and digital mixing consoles, the DAT-56 provides 24-bit AES/EBU and S/PDIF Digital I/O plus provisions for an optical interface. Featuring a SCSI interface for real-time digital recording, DSPnet, and the power of Motorola's 27 MHz DSP56001, the DAT-56 sets the standard for PC-based Digital Audio DSP applications.

SDI DEPLOYED INTO THE NINETIES

Ariel's Signal to Disk Interface is the most flexible system available for recording, editing and playing back of high fidelity audio. Since SDI uses dedicated SCSI disks for recording and playback, it's guaranteed to work in any PC, regardless of the PC's speed, memory capacity, disk controller, etc. Case in point, programs for real-time recording to the PC's disk are now bundled with virtually all of our DSP development systems. If you have a fast PC with a large, fast hard disk, this may be OK. But, since proper operation is dependent on system software such as DOS, the BIOS, TSR programs, and esoterica such as disk interleave and fragmentation, we cannot guarantee flawless performance at high sample rates. SDI, on the other hand, bypasses all of these concerns and offers a complete file management system and editing facility to boot!

600 MBYTE SUPPORT FOR SDI

Conventional magnetic disk drives continue to grow in capacity and price/performance, so, we now offer a 600-Megabyte disk drive in addition to the current 80- and 300-Megabyte versions of SDI. The rule of thumb for calculating SDI recording time is that you need 10 Mbytes for each minute of full-bandwidth stereo audio. For example, the new 600 Mbyte SDI disks stores 60 minutes of two channels at 50 kHz sample rate. That's two hours of single channel recording, and proportionately longer recording times for lower sample rates.

SDI-600I is the SDI system with a nonremovable 600 Mbyte disk drive with cabling and hardware for internal mounting inside your PC's case. SDI-600E includes the 600 Mbyte disk in an external case with 110/220V power supply and all necessary cabling. To add this drive to existing SDI systems, order EXT-600 for the drive plus enclosure and power supply.

OPTICAL SUPPORT FOR SDI

SDI now supports the Sony 600 Mbyte read/write optical disk. Ariel now offers the SDI-ODE, which is the SDI system plus the optical drive and its controller card mounted in an external enclosure with a 110/220V power supply. The optical disk is ideal for archiving large amounts of data, for sound libraries or for applications where data portability is important. The current release of SDI supports R/W optical disks just like conventional disks, with the addition of a new menu to handle loading and unloading of optical cartridges. The optical drive alone, EXT-ODE, can be added to any SDI system, now or in the future.

THE NEW SYSid

Ariel's SYSid Acoustic Test System has been updated to include a new menu-driven user interface. Developed in conjunction with AT&T Bell Labs, SYSid transforms your IBM PC or compatible into an inexpensive, accurate and complete Acoustic Test Instrument. For a limited time, Ariel will include with each SYSid system purchased our Audio ToolKit, consisting of a set of audio processing utilities (recording/playback, real-time FFT, oscilloscope, etc.) for the DSP-16 Plus. Call for a copy of our SYSid demo disk and see why SYSid is your ideal acoustic measurement solution.

RAM PRICES TUMBLE

RAM prices have come down and we're happy (well, willing at least) to pass the savings on to you by reducing the price of our SRAM upgrades. A typical example is the price of our 64K word upgrade for the DSP-56 (64U-56), which has been reduced from $595 to only $300.

$500 PRICE REDUCTION

The cost of DSP chips has also come down (due to market factors plus our increased volume) and, once again, we have decided to pass these price reductions on to our customers. Combining the decreased price of SRAM with the lower cost of DSP chips we are lowering the price of both the Ariel DSP-16+ and DSP-32C by $500 to $1995 and $3495 respectively. For those who already own a DSP-16+ with the PDS-320 development software, we have even a better deal. For a limited time, we are offering the DSP-16+ unbundled for only $1495 ($1795 with the C25 option). Now that is a good deal!

GETTING BETTER ALL THE TIME

Ariel, in its never-ending quest to squeeze every last bit out of its analog I/O, is pleased to announce the final characterization of the DSP-56 and DSP-32C analog sections. Our boards are widely recognized as offering the highest quality audio conversion available for the PC. How do we accomplish this? With great difficulty! Analog design can be treacherous under the best of circumstances and the PC environment is particularly nasty. A recent advance in conversion technology, the oversampling ADC has improved matters considerably. Gone are concerns such as sample-and-hold feedthrough, group delay distortion, and differential nonlinearity. Antialiasing is performed automatically, with near perfect linear phase digital filters. THD+noise is .003% typical, and dynamic range is 90-94 dB flat, 93-97 dB, A-weighted. On the output side, we've greatly improved sample clock feedthrough rejection, increased dynamic range, lowered distortion and added a built-in 20 kHz 9-pole elliptic reconstruction filter. Output THD+noise is a low .003% and output signal-to-noise ratio is 96 dB flat, 100 dB A-weighted. Because the sample clock rejection is so great, the need for an external filter is greatly reduced. In fact, many users have found that the on-board filter provides good results even at sample rates within the filter's passband.

Here are some questions to keep in mind when considering alleged 16-bit analog I/O systems:

Antialiasing filters: Tracking sample rate? Linear phase? Adequate out-of-band rejection? *A 3-pole analog filter provides insufficient protection from aliasing.*
A/D Converter: True 16-bit performance? *Many are barely disguised 14-bit converters!*
Output reconstruction filters: The more poles the better. Are the filters phase compensated? Is there sin(x)/x correction? *If not, the frequency response will roll off and the signal will be phase-distorted.*
Signal-to-noise ratio: The 'acid test' of analog performance. Can the system pass a signal with 90 dB dynamic range? Note that many manufacturers measure dynamic range by writing a zero to the DAC and then measuring output noise. This technique completely eliminates any noise contribution from the A to D. This is cheating! *If the difference between a*

full scale signal and the smallest signal resolvable is not at least 90dB, you're not getting 16 bits!

Needless to say, Ariel's high-end audio DSP boards meet these criteria. If you require the highest performance DSP and the highest quality analog I/O, Ariel has a DSP system for you.

AND THE WINNER IS...

The response to our Fall Newsletter's free giveaway of Ariel Polo shirts has been overwhelming. With dozens of DSP applications articles submitted, it's unlikely we'll be able to print them all. Thanks to all who contributed! We enjoyed reading each and every one. As promised, we held a random drawing and the lucky winner of the Sony portable CD is Dan Greenwood of Northern Illinois University whose application appears below.

WHAT ARE PEOPLE DOING WITH THIS STUFF, ANYHOW?

Dan Greenwood of Northern Illinois University reports that Northern has implemented a powerful image decompression routine using fractals on the Ariel PC-32C card. Results show that by using the PC-32C, the decompression routine is capable of executing over 50 times faster than with a 16 MHz 386/387 system alone.

On the other side of the Atlantic, Gert Rosenboom of Stage Accompany BV, The Netherlands, is exploring the field of fractal audio on the PC-56. Based on articles published in the October 1989 AES Journal, Gert is attempting to hear sounds no man has heard before. Gert has also developed a fractal graphic generation program based on the Mandelbrot fractal set using the PC-56 as a coprocessor. The program generates a fractal image in under 50 seconds. That's fast! If any of you are interested in a copy of Gert's fractal program, it is available on our DSP BBS.

THIRD PARTY NEWS

Hyperception, Dallas TX, has announced that their Hypersignal-workstation software is now compatible with Ariel's DSP-56, PC-56 and ADC56000 boards. Hypersignal workstation is a digital signal processing package containing various real-time signal processing functions including time domain display and waveform editing, frequency domain display and power spectra estimation. Contact Hyperception at (214) 343-8252.

Momentum Data Systems, Costa Mesa CA, has released filter code generators for Ariel's DSP-16+, PC-C25, DSP-56, PC-56 and DSP-32C boards to be used in conjunction with their FDAS Filter design packages. Both the DSP-16+ and DSP-56 versions support single channel or dual channel operation. For further information and a free demo disk call Momentum Data Systems at (714) 557-6884.

Scientific Conversion, San Francisco CA, has announced a sale on demo and prototype versions of their high-quality FDB-20 Filter Daughter Board for the DSP-16+. This add-on enhances the four DSP-16+ filters to state-of-the-art "Brick Wall" 9th order elliptics with group delay compensation, pre/de-emphasis, and DC offset correction. For a limited time you can add "beyond" studio-quality capabilities to your present DSP-16+ or next purchase of an Ariel DSP-16+ at a discount price. Contact: Scientific Conversion at (415) 821-6464.

BITS & BYTES

- **They're here.** A set of C-language drivers for the PC-56, PC-56D and DSP-56 well as a SCSI driver for the DSP-56 and DAT-56 is now available. Contact the factory or BBS for a free copy.
- **Record and Play**. Also available on the BBS is a program that allows the DSP-56, PC-56D and PC-56 to record directly to the PC's internal hard disk. Stereo sampling rates up to 50 KHz have been obtained with fast hard disks using the DSP-56.
- **Turbo DSP56001.** The new 27 MHz version of Motorola's DSP56001 is now available as an upgrade to Ariel's DSP-56 and PC-56 for only $150. Contact the factory for more details.
- **Bug upgrade.** Version 1.51 of Bug-56 is now available. New features include complete DSP-56 support, full speed breakpoint event counts, and host port redirection to/from files.

▪ ▪ SHOW CALENDAR ▪ ▪

Sept. 10-15 ICMC, Glasgow, Scotland

Sept 21-25 AES, Los Angeles, CA

Oct 16-18 BUSCON, Boston, MA

Nov 16-19 ASHE, Seattle, WA

Ariel

PRODUCT INQUIRY FORM

If you have any questions or you want to learn more about our products, please return this form.

Please send more information on:

[ ]	**MM-96**	Motorola 96002 Based DSP Development System
[ ]	**S-56/S-56X**	DSP56001 Based DSP Cards for the SUN Sbus
[ ]	**PC-56D**	PC-56 With Built-in Preamps and NeXT DSP Port
[ ]	**DAT-56**	DSP56001 Based DSP with AES/EBU and S/PDIF I/O
[ ]	**SpeechStation**	Complete Speech Analysis Package for the IBM-PC
[ ]	**DSP-16 +**	New, improved TMS320 based DSP and 16 bit analog I/O for IBM PC
[ ]	**DSP-32C**	AT&T's DSP-32C based DSP and pro-audio quality analog I/O
[ ]	**DSP-300**	Digital Signal Coprocessor with FFT's for HP 200/300
[ ]	**DSP-56**	DSP56001-Based Data Acquisition Processor for IBM PC
[ ]	**PC-56**	Low Cost DSP56001 Coprocessor Card for IBM PC
[ ]	**PC-C25**	Low Cost TMS320C25 Coprocessor Card for IBM PC
[ ]	**PC-32C/M**	Low Cost DSP-32C Coprocessor Card for IBM
[ ]	**FDB-20**	Brickwall filters for DSP-16 PLUS
[ ]	**BUG-56**	Advanced Monitor/Debugger for Ariel's DSP56001 Cards
[ ]	**D3EMU**	Advanced Monitor/Debugger for Ariel's DSP-32C Cards
[ ]	**SYSid**	Acoustic Test System for IBM PC
[ ]	**SDI**	Signal to Disk Interface Recording System for IBM PC
[ ]	**FFT-320**	FFT Subroutines Package for DSP-16 and DSP-C25
[ ]	**ADC56000**	Analog I/O for Motorola DSP56000ADS Development System
[ ]	**DM-N**	All-Digital Microphone For NeXT and Compatible DSP Ports

[ ] **Please have an engineer call to discuss my application.**

Name: ______________________ **Phone:** ______________________

Company: ____

Address: ____

City: ______________________ **State:** __________ **Zip:** __________

Aplication: ____

____

Comments: ____

____

Ariel Corporation ■ 433 River Road ■ Highland Park, NJ 08904 ■ Telephone: 201-249-2900
Telex: 4997279 ARIEL ■ Fax: 201-249-2123 ■ DSP BBS: 201-249-2124 (300 - 9600 bps)

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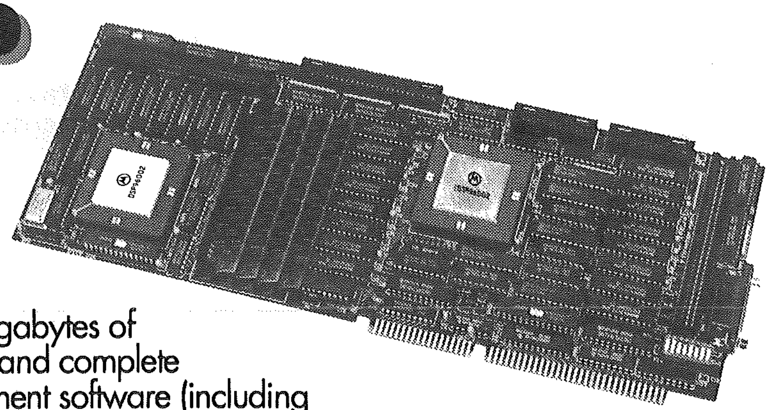
100 MFLOP ENGINE



Based on Motorola's 50 MFLOP 96002, Ariel's Dual DSP MM-96 blasts through real time signal processing, graphics, floating point number crunching and multimedia applications like nothing else.

The MM-96 hooks directly to frame grabber cards via its DT-Connect™ interface and to digital audio with Ariel's DSPnet™ multimaster bus.

Configurations for IBM AT compatibles are available with up to 16 megabytes of memory and complete development software (including an optimizing C compiler, host drivers, and demo software).

Ariel provides the best applications support in the business via telephone, mail, fax, or our 24 hour DSP BBS.

The MM-96 is available now. Call for 96002 support on other platforms.

Ariel_

Ariel Corporation
433 River Road
Highland Park, NJ 08904
Telephone: (201) 249-2900
Fax: (201) 249-2123
DSP BBS: (201) 249-2124

The Ariel Digital Microphone represents a breakthrough in microphone technology—analog to digital conversion in the palm of your hand. It's the perfect NeXT™ accessory, complementing the computer's heralded signal processing and sound generation capabilities to make possible applications from speech-recognition to multimedia presentations.

Specifications

Programmable Sampling Rates	88.2 KHz 22.05 KHz 44.1 KHz 11.024 KHz 5512 Hz
A/D Dynamic Range	92 dB
Total harmonic Distortion (@ 1 KHZ)	0.005%
A/D integral filter in-band ripple	<0.003 dB
Frequency Response (direct inputs)	DC to 42% of Sample rate -.5db
Input connector	mini-stereo (one per channel)
Input type	Differential or single ended
Input gain (contiguously adjustable)	1 to 10X
Minimum input for full scale reading	0.6 V peak to peak
Maximum input without clipping	6.0 V peak to peak
Input impedance	10 kOhms (differential) 5 kOhms (single ended)
Power Supply	Supplied by NeXT DSP Port
Cable length	2.5 m
Size	16 cm x 2.5 cm x 3.7 cm

For further information, contact:

Ariel_

Ariel Corporation
433 River Road
Highland Park, NJ 08904
201/249-2900
201/249-2123 FAX
201/249-2124 BBS

Ariel_

Model DM-N

Ariel Digital Microphone

A Leap Forward in Microphone Technology

The Ariel Digital Microphone is the world's first microphone based wholly on digital electronics. It's sleek, compact case contains all the components needed to convert sound waves to a digital data stream. By converting analog sounds at the source, the signal degradation inherent in analog transmission is eliminated. The digitally transmitted information is virtually impervious to contamination in even the harshest environments. Interference (like hum and buzz from power amps and CRTs, respectively), which has plagued traditional analog microphony, is now history.The DM-N uses two 16 blt Motorola 56ADC analog-to-digital converters delivering a dynamic range of greater than 90 decibels (compact disk quality) with total harmonic distortion of less than 0.005%.

The Ariel Microphone— A Data Acquisition Instrument

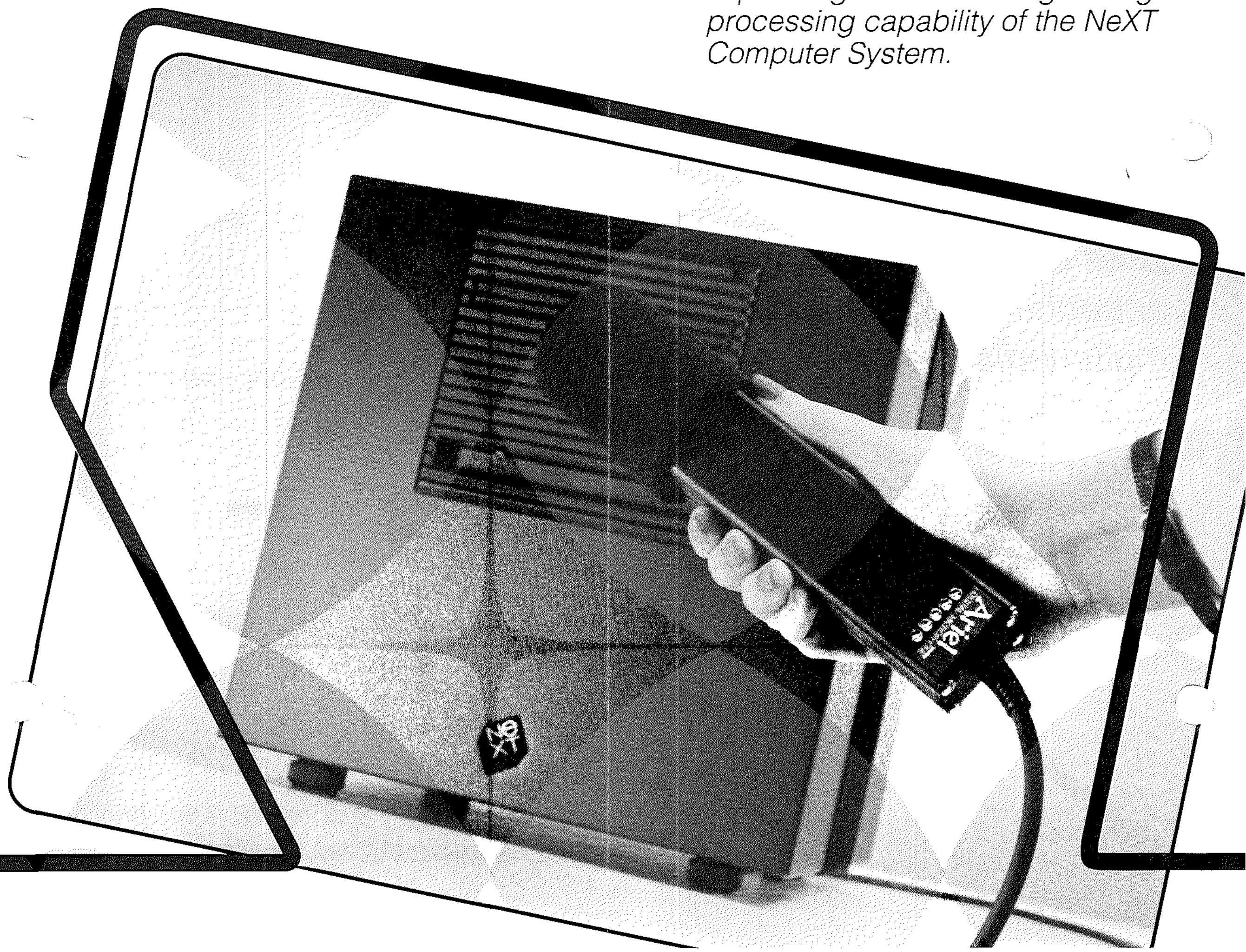
Applications for the Ariel Digital Microphone go far beyond voice recording. External signals from CD players, FM receivers, or other audio sources can be connected to a pair of rear-panel mini-phone input jacks. For instrumentation and industrial control applications, this external input capability can be used to convert signals from a wide variety of sources, from accelerometers to thermocouples. Each input can accept either single-ended or differential signals and the DM-N's sample rate can be selected (via software control) to match the bandwidth of the signal of interest, from a maximum rate of 88.2 kHz down to 5.5 kHz, otimizing storage space on the NeXT disk drive.

The Ariel Microphone— A Multimedia Instrument

Connecting an external source to either rear panel jack disables the corresponding channel's microphone element. With a single external source connected, the remaining channel's microphone continues to operate. This unique feature allows pre-recorded material and voice to be combined in real-time, greatly facilitating the creation of multimedia presentations.

Total Compatibility with the NeXT Computer

The Ariel Digital Microphone is compatible with all existing software that makes use of the NeXT™ DSP port, including "sndrecord", "sndplay", and "MonsterScope". Supplied with a removable windscreen, 2.5 meter cable, and users manual, the Ariel Digital Microphone comes complete with everything needed to turn your NeXT computer into a powerful digital audio workstation.



Since 1982, Ariel has been a supplier of advanced DSP development tools and DSP hardware for the IBM PC, Hewlett-Packard workstations and now the NeXT Computer. BUG-56, a symbolic debugger for the Motorola DSP56001 DSP chip is bundled with the NeXT Computer and is available for other platforms. The DM-N Digital Microphone developed for NeXT™ compatability will also be made available in other protocols and digital formats. Ariel plans to introduce a number of NeXT Bus- and DSP Port-based hardware products expanding and enhancing the signal processing capability of the NeXT Computer System.

THE COMPLETE ACOUSTIC TEST INSTRUMENT

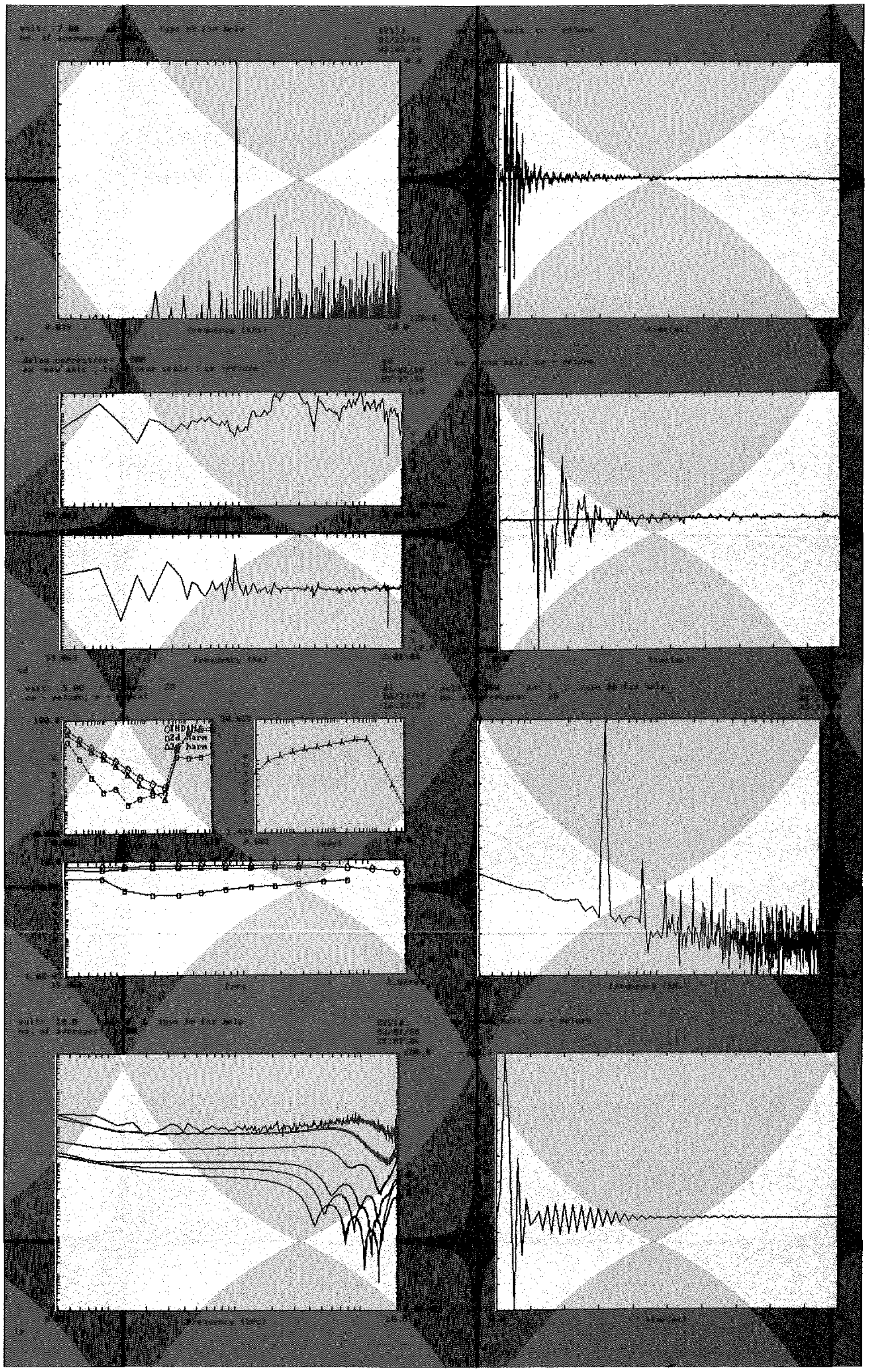
- System Characterization with IBM PC or Compatible
- Wide Range of Narrowband and Broadband Tests
- Accurate Results in Seconds
- Dual Channel Analysis, 16 bit PCM Analog Subsystem
- Full Audio Bandwidth (up to 50 kHz Sample Rate)
- Easily Interpreted Graphic Displays
- Save Results to Disk for Future Comparisons
- Fast Hard Copy with Laser or Impact Printer
- Simple Two Letter Commands for All Functions
- Developed and Used at AT&T Bell Labs

SYSid™

Ariel

SYSid™

THE COMPLETE ACOUSTIC TEST INSTRUMENT



SYSid

SYSid (for SYStem IDentification) software was developed over an eight year period by acousticians at AT&T's Bell Labs for their own use and has now been ported to the Ariel DSP-16, a 16 bit, 2 channel signal processing peripheral for the IBM PC.

SYSid represents a significant step in the evolution of test instruments. It combines high quality signal acquisition and synthesis, a number-crunching DSP chip, a personal computer, and an acoustician's expertise to create a complete, affordable test and measurement instrument for characterizing any system operating within the audio bandwidth.

SYSid delivers the capabilities of systems costing many times its price, performing traditionally time consuming measurements quickly and without sacrificing accuracy.

HOW IT WORKS

SYSid excites a system with a stimulus, either narrowband (sinusoidal) or broadband (chirp or pulse). It then records, analyzes and displays the system's response. SYSid can sample and analyze two channels simultaneously, preserving their phase relationship.

SYSid is versatile. It allows the fast and easy measurement of frequency response, phase response, noise floor, group delay, distortion, impulse response, signal averaging, third octave analysis, linearity, THD and other characteristics.

SYSid employs advanced signal processing techniques, to make the measurements robust. The system will reduce even high levels of ambient noise to elicit the right answer under extreme test conditions.

SYSid BUILDS ACCURACY FIVE WAYS

High Quality Conversion

Real-time 16 bit analog conversion on two channels at sample rates of up to 50 kHz is provided by Ariel's DSP-16 single-card Data Acquisition Processor.

Low Crest Factor Stimulus

Conventional transfer function measurements are made by exciting a system with an impulse and computing the FFT of the result. The impulse can be a poor choice of stimulus for two reasons: high crest factor and transient induced non-linearities.

SYSid offers a more efficient stimulus, the "chirp". A chirp greatly increases the amount of energy delivered to the system per unit time enhancing the signal to noise ratio and reducing nonlinearities.

Time Domain Averaging

Time domain averaging is a powerful technique for increasing a measurement system's signal to noise ratio. SYSid repetitively excites the system and synchronously averages the result. Assuming a time-invariant system, the component of the system response due to the excitation will be reinforced with each average period while all other system noise is reduced. (This includes any uncorrelated noise generated by the electronics of the test instrument.) The result is honest-to-goodness increased dynamic range, by 3 dB for every doubling of the number of averages.

Floating Point FFT Computation

The selectable length, precise FFT, on sizes from 64 to 2048 points (4096 for a single channel), is computed on the PC's 80x87 math co-processor.

Proper Measurement Techniques

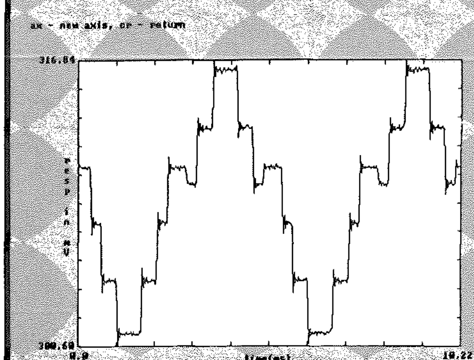
SYSid permits the system to 'settle' to steady state before accumulating its response, thereby eliminating transient artifacts which will otherwise invalidate the results.

APPLICATIONS

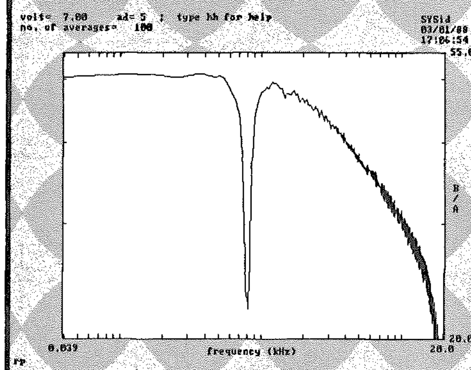
SYSid can be used for applications which range from pyscho- acoustic research to electrical impedance characterization.

Some of these are:

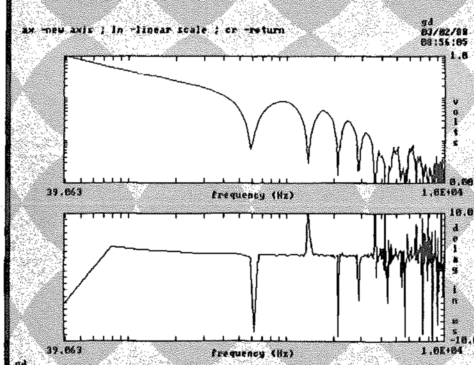
- Transducer testing/evaluation
- Impedance measurements
- Room acoustics
- Servo loop analysis
- Auditory research
- Underwater sound
- Audio performance assurance
- Tape recorder testing/alignment
- Filter analysis & design
- Signal averaging
- Transmission line characterization
- Distortion & noise measurements



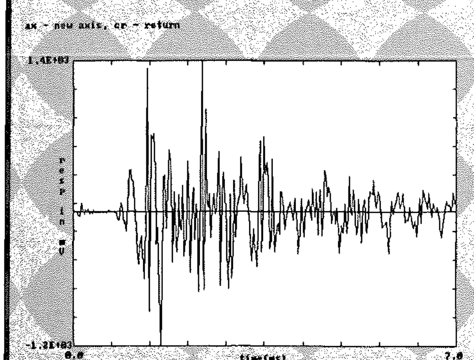
Analog to Digital Converter Time Response



Driving Point Impedance of a 2 Inch Speaker



Response & Group Delay of an ADM System

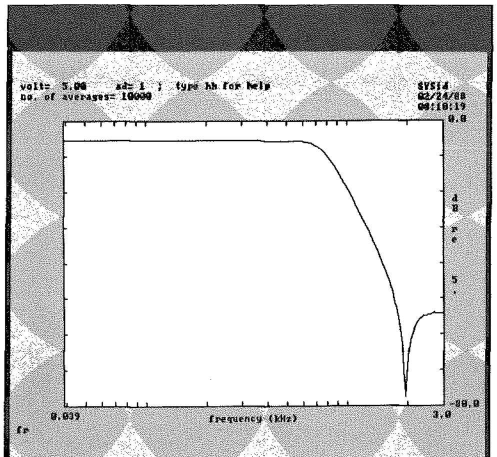


Impulse Response of a Headphone/Microphone System

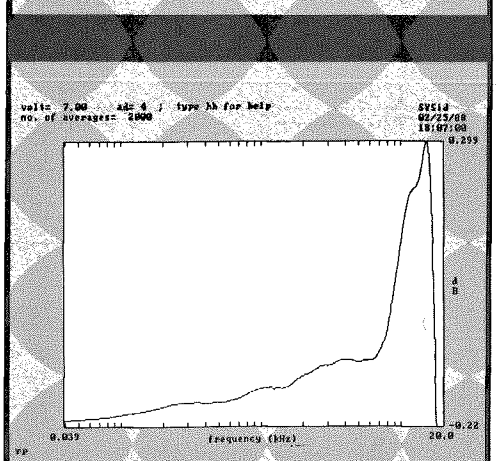
SYSid FEATURES

Transfer Function Analysis

The user can select stimulus type (chirp or impulse), stimulus level, number of time domain averages, maximum frequency analyzed, and transform size (length of FFT). The availability of two independent input channels allows simultaneous transfer function measurements of either channel or the ratio of the two channels. Ratiometric transforms permit deconvolution, impedance measurement and servo loop analysis. SYSid plots magnitude and phase (or group delay) response. A delay correction factor can be selected to compensate for a constant system time delay.



Filter Transfer Function



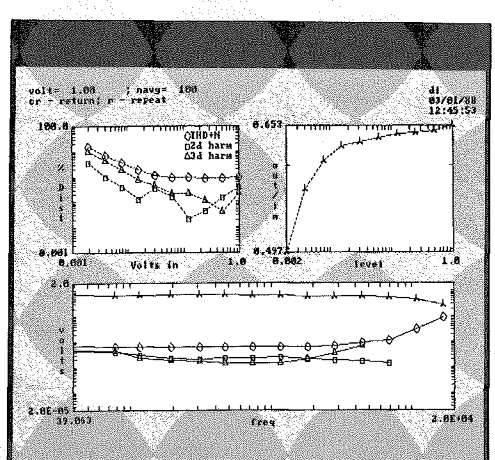
Passband Detail of a Low-Pass Filter

Normalized Transfer Function

Transfer function plots can be normalized by a previously stored response file. The file name used for normalization is displayed on the screen.

Distortion

SYSid provides automated measurement and display of both harmonic and intermodulation distortion. SYSid computes and displays distortion vs. frequency AND distortion vs. level. When measuring THD, three distortion traces are plotted; second harmonic, third harmonic and total harmonic distortion plus noise.



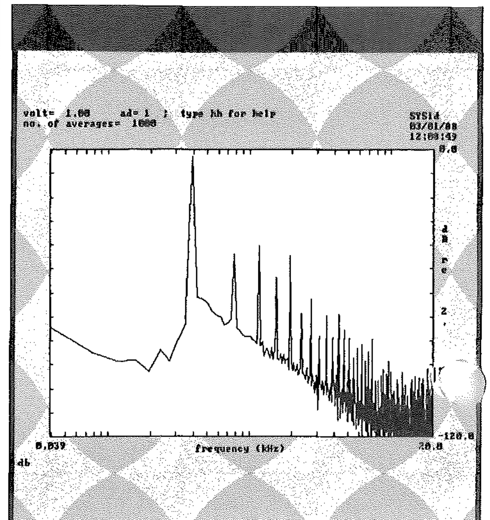
Tape Recorder Distortion

Third Octave Bands

SYSid plots the spectral energy in each 1/3 octave by averaging all frequency bins in the band.

Narrowband Analysis

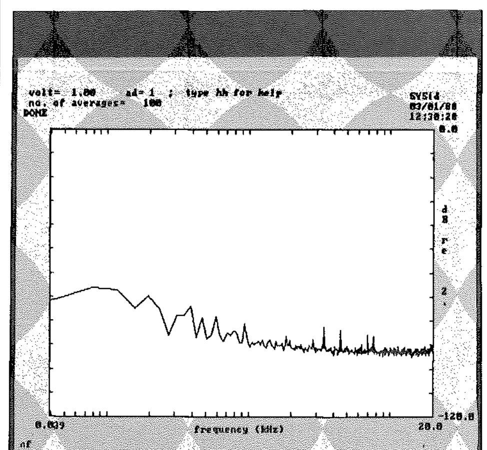
SYSid permits pure tone (sinusoidal) stimulus at user selectable frequency and amplitude for analyzing system response to pure tones. Narrowband analysis is useful in many measurement situations ranging from evaluating analog to digital and digital to analog converters to non-linear system characterization.



Pure Tone Analysis of a Compressor/Expander

Noise Floor

With no stimulus applied, the system's output (averaged in the time domain) is repeatedly transformed and a spectral ensemble (RMS) average is computed by SYSid. Ensemble averaging continues until interrupted by the user.



Noise Floor of an Analog Tape Recorder

Graphic Display of Results

SYSid graphs the results of all functions. All pertinent measurement parameters, i.e. maximum frequency, input mode, etc., as well as time and date, are automatically logged on the screen. Comments can also be added to the screen. The user can select the following plot attributes:

- LOG/LIN frequency.

- Plot limits.

- LIN/dB amplitude scale.

Disk Storage of Results

The system response can be saved in a data file for further use or examination. SYSid data files can be normalized by one another.

Hardcopy of Results

A resident utility is provided so SYSid results can be dumped to printer (dot matrix or laser) for a permanent record in seconds. The printout includes all pertinent information such as date, time, input mode, number of averages, and user comments.

Selectable Length, Precise FFT

Floating point FFTs are computed using an 80 x 87 math co-processor. Transform lengths from 64 to 2048 points (4096 points for single channel) are supported. Future releases will support transform lengths up to 32K points.

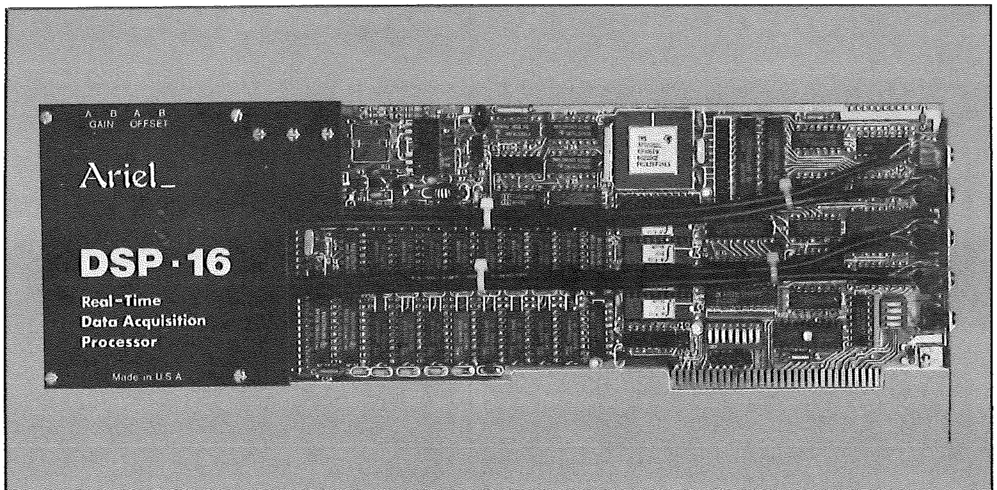
Time Averaging

Averages are calculated with 32 bits of precision. The user can select the number of signal frames averaged, from 1 to 32,767.

Defaults to Disk

All current SYSid parameters can be saved to disk for later recall, permitting easy setup of automated test procedures.

HARDWARE SPECIFICATIONS
The DSP-16 Digital Signal Processor



Two Analog Inputs, Concurrent Sampling

The two analog signals are low-pass filtered, to prevent aliasing, and sampled simultaneously, thereby preserving their phase relationship. Inputs are electronically balanced with infinitely adjustable gain and offset. All connections are via mini-stereo jacks.

Full Audio Bandwidth, up to 50kHz Sample Rate

The sample rate is user selectable from 1kHz to 50kHz.

High-Precision Conversion, 16 bit ADC/DAC

The high quality analog to digital and digital to analog converters deliver up to 90 dB dynamic range.

Digital Specifications

The DSP-16 uses a 20 MHz TMS32020 (TMS320C25 optionally) processor with 32 Kbytes of no wait-state RAM and 256 Kwords dynamic buffer RAM.

Other DSP-16 Based Systems

The DSP-16 hardware is also available as part of a complete TMS32020 or TMS320C25 Program Development System, or as a part of Ariel's SDI Signal-to-Disk Interface, which allows real-time data acquisition direct to high capacity SCSI disk.

More standalone software applications for the DSP-16 will be available in the near future. Contact Ariel for more information.

SOFTWARE COMMANDS

All SYSid functions are accessible by the following two letter commands:

ad - modify AD channels for frequency response
ax - replot the plotting window and header info
co - add screen comment
da - change the d/a channel (0,1)
db - toggle the plot between voltage and db
de - change the phase delay correction factor
di - calculate the system distortion
fr - measure the frequency response using sweep
gd - calculate the group delay
hh - display this message
ip - impulse response
mf - change maximum frequency
na - modify the number of averages taken
nf - record the background noise floor with no stimulus
nm - normalize one SYSid data file by another
oc - compute third octave band results
ph - calculate the phase response
pl - change plotting limits
qu - quit SYSid
rd - read in default conditions
re - read in an old SYSid data file
rp - replot last plot
sa - save the last response to disk
ti - plot time response
tl - change the transform length (128,256,512,1024, 2048, 4096)
to - stimulate and record a pure tone
ts - plot the stimulus time response
vo - modify the output voltage
wd - write present defaults to disk

SYSTEM REQUIREMENTS

- IBM PC, XT, AT or compatible with 448K or more
- One or more double sided floppy disk drives
- DOS 2.0 or later
- 1 PC compatible expansion slot
- AT&T, EGA, CGA, or Hercules graphics
- 8087, 80287, or 80387 Math Co-processor

PACKAGING

SYSid consists of a DSP-16 Data Acquisition & Signal Processing Board, SYSid Software, and an Operating Manual. Manuals can be purchased separately and their cost can later be applied toward the cost of the system. SYSid can also be provided as an upgrade to existing DSP-16 boards.

WARRANTY

The DSP-16 hardware has a full one year warranty for defects in workmanship or materials.

Ariel

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