MARS

MUSICAL AUDIO RESEARCH STATION

The Musical Audio Research Station (MARS) is a specialized digital machine for real-time audio processing applications developed by IRIS, the research institute of FARFISA, as an integrated environment in which a graphical user interface, an embedded real-time operating system and two powerful and fully programmable IRIS digital audio processors (X20) are linked together to create a universal sound processing machine.
A proprietary development system, the IRIS EDIT20 software package, provides musicians a graphic user-friendly interactive tool for programming and performing, through the connection and the immediate sound feedback of audio "objects" coming from an expandable library to build every kind of patch and DSP algorithms for musical/signal processing such as sampling, analysis and synthesis, filters and sound effects.

Several original applications are available already: LPC analysis and synthesis, Vocoder, physical models, graphic FFT... and many others can be developed by C-programmers using APPLI20 toolkit. Some works of Computer Music have been produced (S. Sciarrino, L. Berio, W. Prati).
Real-time digital audio multi-algorithm processing, programmable links between MIDI controls and physical algorithm parameters, dynamic allocation for timbres and voices, make the MARS a powerful interactive and integrated environment to be used for audio research, musical production and Computer Music pedagogy, and, once configured, as a musical MIDI instrument that can be played by using sequencers, MIDI devices or programming language such as MAX. MARS is actually interfaced with standard MIDI devices and the Atari ST and TT computers via the VME bus.

IRIS

MARS SPECIFICATIONS

DSP's CHARACTERISTICS

Pipelined architecture, Fully programmable
2's complement fixed point arithmetic on 24 bits
16x16 bits multiplyer, 24x24 bits Arithmetic and Logic Unit
768x24 bits words of internal data memory, 512x64 bits word of external program memory
40 MHz System Clock, 39.0625 kHz up to 1 MHz of Sampling Rate

EXAMPLES OF SOME DSP PERFORMANCES*

128 second order filters
128 multi-segment envelope generators (always available)
256 fully controllable simple table-lookup oscillators
128 fully controllable interpolated oscillators
128 delay lines with maximum delay of 7 minutes
32 independent harmonizers
2 independent 2048 point FFTs
32 voices of Karplus-Strong string algorithm
32 voices of 4 operators FM algorithm
2 voices of 25 cells LPC algorithm

AUDIO LINES

4 Mono Input Lines TRS balanced jack-type	CRYSTAL Delta-Sigma A/D converters Low Noise and Distorsion > 90 dB S/(N+D) Internal 64 x Oversampling Linear Phase Digital Anti-Alias filtering 39.0625 kHz as Standard Sampling Rate
8 Mono Output Lines TRS balanced jack-type	CRYSTAL 64x Delta-Sigma D/A converters 97 dB Dynamic Range over the Audio Band A weighted Internal 8x Oversampling Linear Phase Filter 39.0625 kHz as Standard Sampling Rate
2 Stereo Headphone Outputs	High Impedance (600 Ohm)

STANDARD CONFIGURATION OF MEMORY CARDS (16 bits word)

2x1 up to 2x8 Mwords of DRAM SIMM modules NEC MC-421000A8 MC-424100A8 (80ns)	7 minutes of sounds or delay lines
2x256 Kwords up to 2x2 Mwords of EPROM EPROM sockets for INTEL 27C010/20/40/80 (120ns)	1.75 minutes of preset samples

SM1000 AND HOST COMMUNICATIONS

1 MIDI interface	IN, OUT, THRU
1 RS232 interface	19200 Baud
1 VME interface	500 Kbytes/sec maximum transfer rate

SOFTWARE PACKAGE

RT20M	Embedded Real Time Operating System
EDIT20	Interactive Graphical Development Environment
APPLI20	Application toolkit

OTHER SPECIFICATIONS

Power requirement	100-240 v AC, 50-60 Hz
Dimensions	2U Standard Rack Module

* algorithms realized with optimized assembly language

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