I'M EASY

Audio Processing and Sound Spatialization effects in Real-time

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Integrated Multiprocessor Expandable Audio Spatialization System

I'M EASY

An activity within TETRApc TTN, ESPRIT IV HPCN

I'M EASY is an interactive and easy to use system to simulate and produce spatial effects in real-time, saving up to 30% of the costs of a manual and/or hardware solution. I'M EASY fills a gap in the current commercial market, as it provides a device for *creating in real-time* original spatialisation efects that cannot be preencoded or prerecorded, and need to be created during the performance of live events, theater applications, interactive museum shows and entertainment parks.

Pro Audio Professionals involved in audio and multimedia installations, post-production services, and in the field of entertainment activities, including game parks, theater and outdoor events can benefit from I'M EASY audio spatialisation system.

To carry out these services, a prototype will be implemented.

I'M EASY è un sistema facile di usare ed interattivo per simulare e produrre effetti di spazializzazione in tempo reale, diminuendo del 30% dei costi di una soluzione manuale e/o hardware. I'M EASY riempie una mancanza nel mercato commerciale, offrendo un dispositivo per *creare in tempo reale* effetti originali di spazializzazione che non possono essere precodificati o preregistrati, e richiedono di essere creati durante la prestazione di spettacoli al vivo, applicazioni di teatro, mostre interattive di musei, e parchi d'intrattenimento.

I professionali nel settore Pro Audio di installazioni audio e multimedia, servizi di post produzione e nelle attività di intrattenimento, includendo parchi di gioco, teatri ed eventi all'aperto, possono trarre beneficio dal sistema di spazializzazione audio I'M EASY.

Per realizzare questi servizi, un prototipo sarà realizzato.

Features

- Audio processing and spatialisation system
- Highly interconnected parallel network of digital signal processor (DSP)
- Advanced control of multi-channel sound
- Creation of spatialisation effects applied to external signals in real-time
- Scalable and re-configurable

Caratteristiche

- Sistema di spazializzazione e processamento di audio
- Rete parallela di processori di segnali digitali (DSP)
- Controllo avanzato di suono multicanale
- Creazione di effetti di spazializzazione applicati a segnali esterni in tempo reale
- Scalabile e riconfigurabile

[HPCN Center at DSI]

Last Updated September 1, 1999

Ultimo aggiornamento 1 Settembre 1999

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I'M EASY consortium

The consortium includes 7 Partners:

RIGEL/ ARTEC Group

Istituto di Ricerca per l'Industria dello Spettacolo

A & G Soluzioni Digitali

Department of Systems and Informatics University of Florence

CESVIT (High Tech Agency)

GOIGEST

RIGEL/ARTEC GROUP: ARTEC GROUP G.E.I.E./E.E.I.G., Belgium - project coordinator and HPCN technology transfer receiver (Firm).
IRIS: I.R.I.S. (Istituto di Ricerca per l'Industria dello Spettacolo), Paliano-Italy (Firm), has designed an innovative device for Digital Signal Processing, fully programmable and specially tailored for algorithms implementation, that has been realized in ASIC technology. The HPCN capability of the device has been used to design a multi-processor board for Sound Distribution and Movement (Spatialization) that can be adapted to many applications. IRIS will provide a modified board and a basic library for Algorithms implementation, bringing its long experience in Sound Synthesis techniques.

DSI: Dipartimento di Sistemi e Informatica (Department of Systems and Informatics), Università degli Studi di Firenze (University of Florence), Firenze, Italy (Technology Provider) Center for HPCN (Prof. Ing. P. Nesi).

CESVIT: CESVIT (High Tech Agency), Firenze, Italy; TETRApc-TTN (Technology. Provider) TTN partner and disseminator.

A&G: A&G Soluzioni Digitali (End-user) is a company structured on an innovative pattern, which sees technicians and musicians work together every day, dealing with audio and video production carried out with computer workstations. A&G is an exclusive distributor in Italy for Apogee Electronics, APB Tools, CEDAR for ProTools, Gallery and INA GRM products, and also an authorized reseller of AVID/Digidesign and other relevant development partners.

GOIGEST (End-user): Goigest, Milano, Italy; partner and end-users.

I'M EASY contact person

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e-mail: nesi@dsi.unifi.it
WWW: http://www.dsi.unifi.it/~nesi

The Reference TTN

For I'M EASY activity, partners have chosen TTN-TETRApc as their ``reference TTN'' that has been proposed by CPR (Consortium Pisa Ricerche, Pisa, Italy), CESVIT and CSM. In particular, CESVIT High-Tech Agency (a nonprofit agency) has been selected by I'M EASY partners as reference TTN-TETRApc site.

It should be noted that CESVIT for TTN-TETRApc activity collaborates with the Department of Systems and Informatics (DSI partner) that provides support about technological and scientific aspects of TTN HPCN. This collaboration is very suitable since the complementary roles of CESVIT and DSI within TTN HPCN have been well-identified. CESVIT has a very high visibility and sensitiveness with respect to the market, for activities such as distribution, advertising, evaluation, etc. (very useful for TTN-TETRApc management). DSI provides know-how in HPCN technologies. According to the cooperation agreement between CESVIT and DSI for the TTN-TETRApc, the scientific responsible of the CESVIT part of TTN-TETRApc is Dr. Eng. P. Nesi of DSI. Please note that he is also the coordinator of the present activity.

Reference to the TTN TETRApc and its related partners:

TTN HPCN TETRApc	Consorzio Pisa Ricerche
TTN HPCN TETRApc	ARTIC

I'M EASY Objectives

The major objective of I'M EASY concerns the viability of HPCN technology within a particular vertical, rich niche of the entertainment market that will have an important impact on the implementation of new concepts in theatres, orchestras, concerts, cinema, creative studios, musicians, schools of music, soloists, and disco-dance, etc. In particular, the benefits derived from the deployment of I'M EASY technology will have an important influence on professionals and industries involved in the audio and sound reproduction business in which no high performance distributed spazialization system for surround music (object sound spatialisation approach) is present. The adaptation of HPCN solutions will provide new functionalities and high performances to such a system reducing cost and the present component complexity.

Main objectives and expected results of I'MEASY project are:

1. adopting HPCN technology to improve performance and to extend capabilities of the existing design of IRIS, in order to reach a satisfying price/performance ratio;
2. introducing HPCN technology for obtaining, on the basis of already available product SPARK of IRIS, a flexible and reconfigurable parallel system for sound processing and spatialization in order to introduce new features in the commercial market. The system has been considered strongly needed for controlling multiple diffusor installations, because conventional systems, based on automated mixers and outboard equipment, can be very difficult to manage, especially with a large number of output channels;
3. deploying HPCN technology to improve performance and to decrease costs of products dedicated to multi-channel sound system management, as compared to existing products and, thus, for covering a wider market;
4. disseminating results at National and European levels.

The opening of a new market for a new product, as I'M EASY, may be interesting for a wide range of potential users and it has very few competing products in the market.

Executive Summary

I'M EASY project forms an activity within the TETRApc TTN. The activity is of type Demonstration (Task 6.23) and has a duration of 18 months.

The I'M EASY project focuses on an audio and sound processing and spatialization system based on a highly interconnected network of digital signal processors, intended for applications in the fields of entertainment and professional music; the device will be completely reconfigurable and will allow the advanced control of a multi-channel sound system to create and handle spatialization effects applied to external signals.

The targeted end-users are professionals involved in audio and multimedia installations, post-production services, and more generally in the field of entertainment activities, including game parks, theater and outdoor events.

Currently there is great interest in multichannel audio systems, but it is mainly focused on coding/decoding techniques and reproduction sound systems; the proposed system intends to fill a gap in the current commercial market, as it provides a device for creating in real-time original spatialization effects that can then be encoded as needed and reproduced on either custom or already installed sound systems.

The goal is to provide an interactive system to allow spatial effects in real-time; this is mandatory for live events, theater applications, interactive museum shows, entertainment parks, as the exact timing and temporal evolution of spatial effects may be dependent on external events and for this reason unpredictable in advance. The spatial effects cannot be preencoded or prerecorded and are required to be created in real-time.

A high-performance parallel computing architecture for sound spatialization has already been developed in the form of a single board prototype by IRIS, the SPARK board. To enter the market of professional audio business a significant improvement of capabilities is needed, mainly regarding multiple and scalable systems interconnections, and in the field of configuration and software control.

I'M EASY Business Aim

The major business aim of the I'M EASY project consists in the realisation of an innovative product based on the adoption of HPCN technology to improve performance and extend the capabilities of the single SPARK board IRIS prototype (already developed) that is unable, with the current conventional technology, to reasonably satisfy the desired price/performance ratio. The exploitation of HPCN technology has the advantage to reduce risks in the migration to a top market segment of products dedicated to multi-channel sound system management.

The strategy behind the I'M EASY initiative is extremely clear:

- Enhance the existing solution with HPCN approach
- Offer new functionality not allowed to existing products currently commercialised by European and US, as well as Japanese industries
- Reduce the ratio performance price/performance dramatically
- Produce patent(s)
- Impose, in this vertical market, an industrial leadership based on a concrete use of HPCN potential

Industrial Benefits

For evaluating the end-user benefits (i.e., post-producers, directors theatres, etc.), it can be estimated that:

- the price of a typical I'M EASY system (hardware and software) will be about 3,5 KECUs. This price is about 40 % less with respect to the other audio and sound reproduction builders,

- a rehearsal can take place any time and for every single song just only because the artist needs, causing a lot of time wasted for new setups. A music production usually keeps 5-6 versions for each song before ``cutting'' the final master. Using I'M EASY leads to save up to 7.200 ECU's per production/album,

- a professional post-production studio (which can also develop audio for multimedia) costs 150 ECU's per hour or 1200 ECU's per day (forfait). In a medium length project of 60 days, 20 of them are dedicated to the mix for surround/spatialisation. A tool like I'M EASY could reduce these costs for about 8.000 ECUs,

- to obtain an acceptable spatialisation on multi-channel systems or a surround project codified on a stereo master, it is nowadays necessary to operate on audio materials through several steps, from the original record until the final master. By taking advantage of I'M EASY technology, not all the steps will be eliminated, but the same sound engineer that takes care of editing and mixing music could save up to 20% of the time spent programming his console or every single channel movement on software systems, with a cost of 120 ECUs per hour, when a serious audio project takes not less than one week for the mixing (8 to 10 hours per day),

- a multimedia project takes not less than 1 month to finish the audio project and a soundtrack for movies takes up to 6 months. By considering that every single scene needs different positioning and movements for the music and sound effects (all of those on multi-channel basis), I'M EASY can make the sound designer saving up to 30% of the production time.

I'M EASY approach

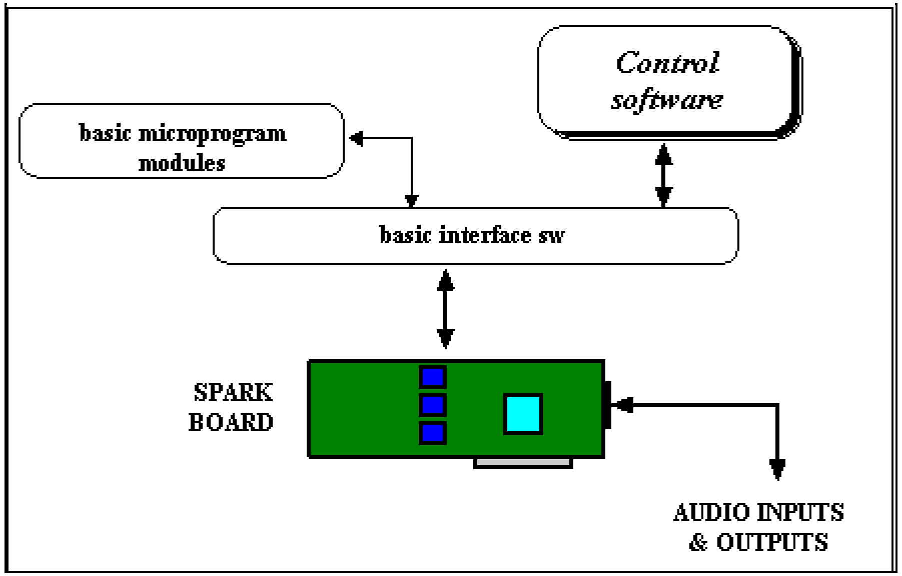
I'M EASY activity plans to exploit the higher computation power, larger on-chip memory, ease of use and generality qualities of the SHARC DSP devices, based on the adoption of SPARK architecture of IRIS.

The adoption of SPARK mainly consists in testing the spatialization algorithms and their validity in a musical context. This will be an efficient support to reduce the effort for implementing a prototype, as well as the interconnection and digital mixing in hardware, and to focus this project on HPCN technology transfer.

A big effort has been made to map as nearly as possible the functionality of the low level libraries for the host computer, which manages all the I'M EASY board resources, to the functionality of the SPARK system. This approach has been chosen in order to maintain the same software architecture successfully tested on SPARK, and to allow testing part of the high level of software on SPARK while waiting for the new board to be developed.
Using HPCN technologies, the complete system will reach a greater value with respect to SPARK, since a new and strongly innovative product with a set of new functionality and higher performance will be implemented.

For these reasons:

- the DSP board will be redesigned starting from the existing SPARK board but with some important additional features: multiple board interconnection, DRAM support, digital mixing section in FPGA, Audio Unit interface.
- the Audio Unit will be independent of the life of the product. The product could be distributed separately even to customers that already own such a unit.
- the Microprogram module set has been partially developed in a preliminary form for the SPARK prototype and has to be redesigned according to the high level specifications.
- the Low-level interface libraries will have to be developed and tested with special care as they are crucial to the correct interaction between hardware and software.
- the Control software will be developed to allow advanced configuration editing, saving and restoring, taking into account the need to dynamically reconfigure the available hardware and firmware resources. The current control software developed for the SPARK board is mainly dedicated to the testing of the simpler functionality and has to be redesigned to handle a reconfigurable environment.



SPARK system: overall architecture

State of the Art

Currently, even if there is a great interest in multichannel audio systems, it is mainly focused on coding/decoding techniques and reproduction sound systems, which are especially important in the cinema and home theater industry.

- Market Situation
- Current Operational Way

The proposed system intends to fill a gap in the current commercial market, as it will provide a device for creating in real-time original spatialization effects that can then be encoded as needed and reproduced on either custom or already installed sound systems.

With I'M EASY activity, the partners have exploited a strongly innovative idea: automating and managing, in real time, the position and the movements in the audio space of one or more audio sources during mixing process and public performance of concerts, as well as for theatre or disco-dance based events.

With respect to other spazialization systems available on the market and to the old architecture of A&G (Sigma 1 plus ProTools), I'M EASY with comparable performance at a lower cost, will increase the number of systems commercialised by A&G with a return of investment, at least of about 45% in three years. The expected cost/price reduction should reach an economy of 40-50 systems compared to other commercial spatialization equipments (not offering the same features of I'M EASY) for a turnkey solution.

I'M EASY general architecture

I'M EASY is a hardware/software solution. The main tasks of 'M EASY are automating and managing, in real time, the position and the movements in the audio space of one or more audio sources during mixing process and public performance.

As the amount of computing power may not be sufficient for applications with many outputs to handle, I'M EASY system will support a multiple-board configuration, where algorithms can be easily partitioned into independent functional blocks. I'M EASY architecture represents a good example of the usage of the HPCN technologies.

Architecture Overview
Involves both hardware and software aspects.

The software architecture description of I'M EASY can be decomposed in three different levels:

- DSP level

  Contains the DSP microprograms.
- Control level

  A set of low level libraries for the host computer, which manages all the EZ-SOUND board resources.
- High Level Control Application (HLCA)

  An interface application program, containing the user interface and the capability to interact with lower software level.

The hardware component of I'M EASY is:

- EZ-SOUND

  A PCI board for PC-compatible computers, to be used in a multi-board assembly by I'M EASY system.

I'M EASY reports

- [Information Document First Semester]

I'M EASY related sites

Organizations

- ASE Audio Engineering Society
- SMPTE Society of Motion Picture and Television Engineers
- APRS Association of Professional Recording Services
- ESTA Entertainment Services & Technology Association
- SPARS Society of Professional Audio Recording Services
- IRMA The International Recording Media Association
- ICMA International Computer Music Association
- NAMM International Music Product Association
- Swedish Studio Engineers Society
- IBS Institute of Broadcast Sound (UK)
- AIMI Associazione di Infomatica Musicale Italiana
- EMF Electronic Music Foundation
- MMA Midi Manufacturers Association
- Italian MIDI
- ATSC Advanced Television Systems Committee
- EBU European Broadcasting Union
- INA Institut National de l'Audiovisuel
- IRCAM Institut de Recherche et Coordination Acoustique / Musique
- ABTT Association of British Theatre Technicians
- ASA Acoustical Society of america
- USITT United States Institute for Theatre Technology
- OISTAT International Association for Scenographers, Theatre Archtitects & Technicians
- Scotts Theatre Links
- BEAST Birmingham ElectroAcoustic Sound Theatre
- IASIG Interactive Audio Special Interest Group
- ACM SIGSOUND Special Interest Group on Sound software and hardware
- AMPAS Association of Motion Picture Arts & Sciences

Spatialization

- Ultimate Spatial Audio Index
- Audio and Three Dimenisional Sound Links
- Euphonia
- Spatial Sound Links

Recording Studios

- Ocean Way Recording
- Logic Recording Studio
- Mulinetti recording studio
- Studio Sintesi
- Capri Digital Studio
- Cat Sound Studio
- Music Lab
- ph Music Work

Journals & Magazines

- Studio Sound
- AudioReview
- Strumenti Musicali
- Audiomedia
- Keyboards
- Studio Post Pro
- Keys
- Computer Music Journal
- Applied Acoustics
- IEEE Transactions on Speech and Audio Processing
- Electronic Musician
- Mix Magazine
- EQ Magazine
- Music & Computers
- Leonardo Music Journal
- Lighting & Sound International - The Entertainment Technology Monthly
- Sound+Communication Systems International
- ACM SIG-SOUND Gopher
- JAC Page at USF
- ACM Special Interest Group on Sound and Computation

Conferences

- International Computer Music Conference
- ICAD International Conference on Auditory Display

Music Files

- Harmony Central Internet Music Resource List
- The "Ever-expanding" Web Music Listing
- The Web Wide World of Music
- A History of Electronic Instruments
- Computer Music Software List
- Archives of Classical MIDI Sequences
- Demonstrations of Renaissance Instruments
- The Binaural Source
- Cyborgasm
- Interactive Sound Installation
- Virtual Audio Sampler

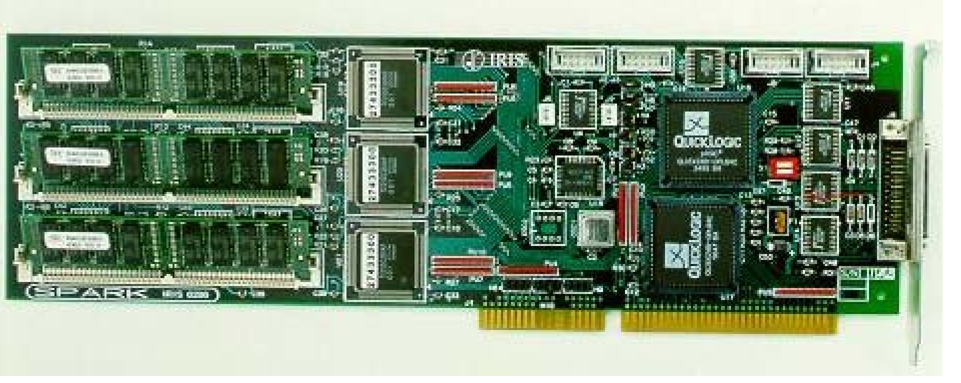
Products

- Apogee Electronics
- Digidesign
- MusicPro - for the music and recording industry
- The Ambisonic
- Dolby Laboratories Inc.
- DTS Digital Theatre Systems
- Opcode
- Origin Records
- Digital Audio Labs
- D.A.W. Mac
- Arboretum Systems
- 3D Sound
- Aureal
- CATT-Acoustic
- Crystal River Engineering
- Firsthand
- Headspace
- Holophonics
- Intel Corp.
- Lake DSP
- Level Control Systems
- NuReality
- Paradigm Simulations Inc.

QSound
- Reality by Design
- Roland Corporation
- Sony
- Spatializer Audio Laboratories
- Hardware/Software

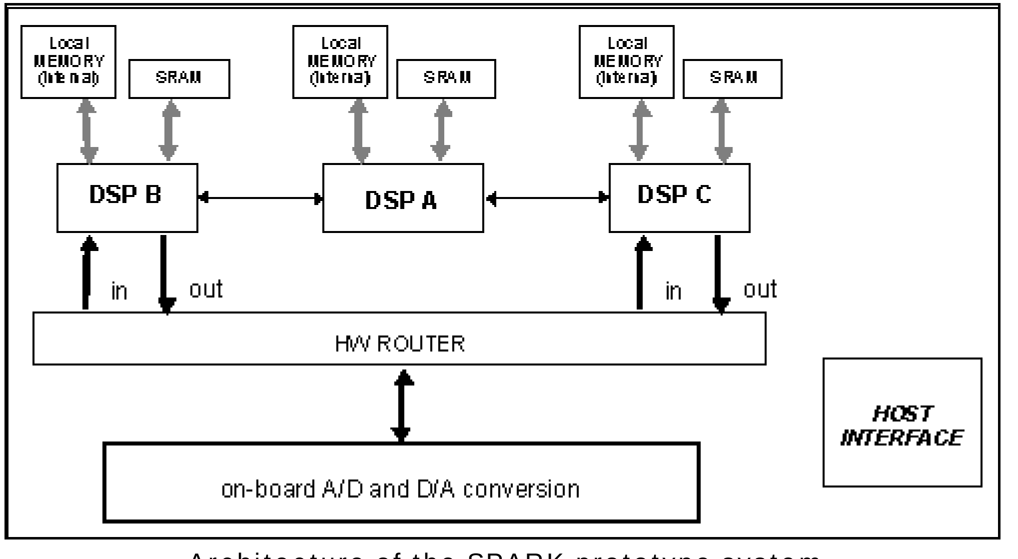
SPARK prototype system

A high-performance parallel computing architecture for sound spatialization has already been developed in the form of a single board prototype by IRIS, the SPARK board. SPARK has already been tested by two selected users, Tempo Reale of Florence (by allowing electronic music composers to use multiple loudspeakers in their realizations), and CRM (Centro Ricerche Musicali) of Rome, to investigate in the field of sound spatialization.



The SPARK system is based on a sinchronous multiprocessor architecture using IRIS custom-designed digital signal processing units. Each DSP board is based on three custom DSPs connected in a synchronous configuration and operating in parallel at full speed. The total aggregate computing power is 150 MIPS (peak) per board, and on-board interconnection bandwidth reaches 26 Mbit/s.

All A/D and D/A conversions are performed on-board. Furthermore, the microprogram assortment and software support is very limited. The SPARK concept has been extensively tested in many situations and it has proved itself very powerful and promising.



Architecture of the SPARK prototype system

Market Situation

Most of the public events see thousand people attending live concerts inside stadiums, theatres, discos where is needed to compensate delays caused by the positioning of the loudspeakers around large areas and to obtain a high level of involvement from the attendance.

Without considering the incredible amount of stadiums as locations for events, every year, in Europe take place about 1000 important live concerts, there are more than 3000 main theatres, and 20 national television networks. Any of them could get benefits by I'M EASY technology.

The Television standard recently migrated to stereo transmissions, where it is possible to encode surround information. It should be noted that producers and directors look more for involving people in high intensity events by using visual effects, which have to get correct positioning inside the three-dimensional audio space. To obtain such a kind of results any technical problem related to the specific area where the event has to be reproduced, must be easily solved by the spatializer system itself.

According to our market analysis, performed by interviewing sound engineers, post-producers directors, musicians, mastering, none of the products on the market is fully satisfactory:

- The existing spazialization and/or surround systems in the market do not match completely the professional expectations like easy installation, management on mixing consoles or on personal computers. None of the products on the market can offer a whole of real-time operations, dynamic automation of each single source from and to the audio matrix and mixer, fully automatable functions controlled by joysticks or MIDI controllers.
- Plug In software like PT3D Spatializer, Dolby Surround Tools and Sigma 1 are related to specific audio cards installed only on Macintosh computers, other stand alone systems like DSS are nowadays obsolete concepts.
- Post-production facilities need to develop their projects on systems matching and even advance the newest industry requirements. DVD and DV-Audio protocols provide for archiving and reproducing surround matrix, or audio on 6 or more channels. According to those new standards post-producers had to rule their investments on appropriate technology.

These are the reasons why I'M EASY will be developed: an exhaustive system for spatialization, real-time automation of every single audio source, dynamically controllable from the major standard controllers and timecodes as requested by important prompters as well the partners in this project GOIGEST, IRIS, RIGEL, A&G and others.

Current Operational Way

There are several hardware and/or software based systems to enlarge stereo images, spatialize, create surround matrix. Most of the them are based on physical commands like buttons or sliders that operate on mixing consoles which still are mixers for audio and not specific products oriented to the audio spatialization.

In these cases the user has to assign one the incoming sources to a channel, then to a send to reach one or more points of a matrix which will redistribute the sound on the desired outputs. The routing and assignation commands involve several steps to reach the desired positioning. In addition to that, even the most recent consoles have only small LCD displays, which force the user to jump from one window to another. These procedures cause a chain of steps not at all quick and comfortable. The less expensive console-carrying feature allowing to program surrounds costs about 2.000 ECUs and does not match exactly the requirements for a handy and fast operative tools.

Actual professional software for 3D, spatialization, and surround are mainly software "Plug Ins" based on Apple Macintosh computers for specific, expensive audio systems. This software offers in general a helpful user interface, are easy to understand and can be driven by mouse and/or commands from the keyboard. The cost of a single application is about 1.500 ECUs, but cannot work without being installed on proper computer based digital audio cards. The necessary hardware for the computer costs around 9.000 ECU. In any cases they partially cover the requirements, in that none of them joins the capabilities of software based product with instantaneous physical controllers.

Moreover, there has never been a spatializer system able to manage several audio sources in real time and, in the meanwhile, to be slaved to synchronisation signals like SMPTE, MTC, and BIPHASE.

Taking into account the above points and considering that:

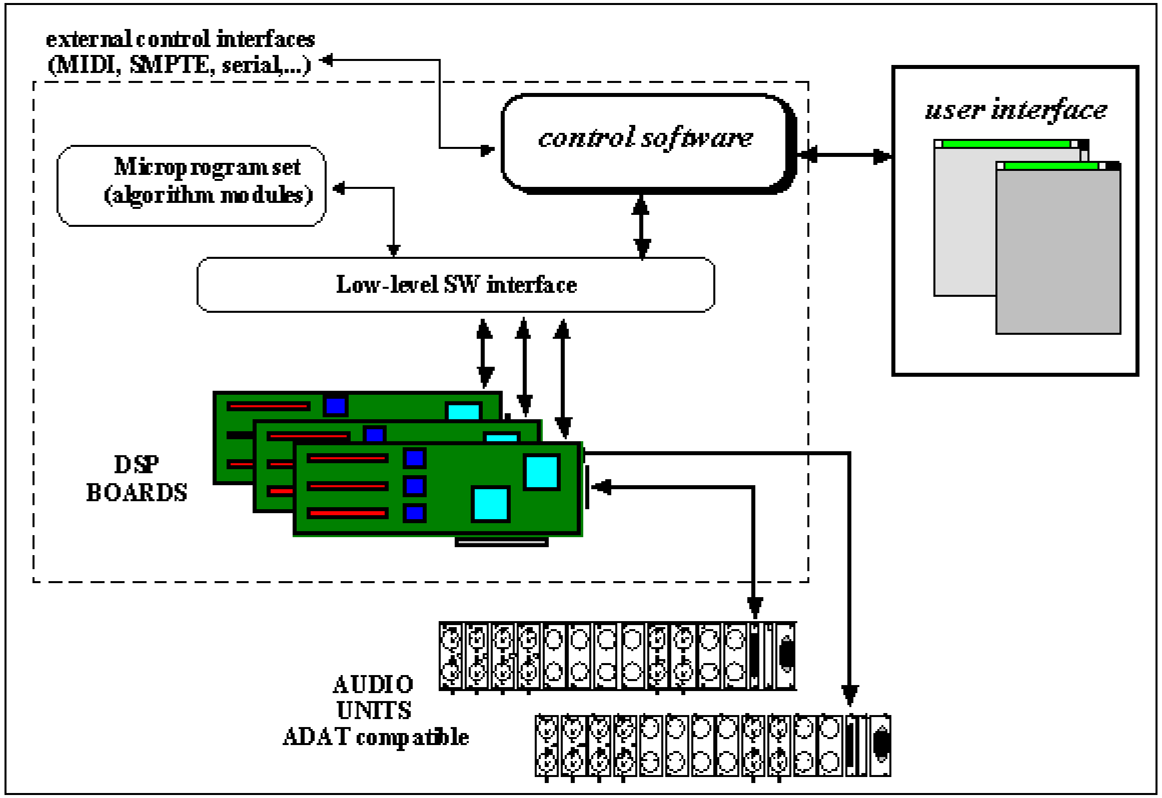
- DSI has already implemented object-oriented music editors and lecterns for both single musicians (LIOO application) and director/archivist (MSLIOO application). Moreover, DSI has experience in decoding MIDI files
- IRIS has the hardware technology related to audio signal processing coming from the realisation in 1989 of the first Italian DSP custom chip, the X20. This technology has been, then, used to generate a family of microchips entirely dedicated to audio processing like the K22, that equipped in the years 1993-1997 the new family of FARFISA professional keyboards (F and G series), and the N22, a new 32 bit fixed point DSP that will be used in the I'M EASY board for sound elaboration and spatialization.
- Rigel Engineering has the know-how and the experience of advanced software development related to multimedia research and innovation. The company also has a strong background on development related to audio processing, for example, within the field of the speech synthesis with the product MyVoice
- A&G and GOIGEST both have the know-how and the practical experience on the field, starting from audio professional business market till the artistic production and realisation of live concerts and theatrical performances.

These facts and the maturity of the HPCN technology produce the opportunity for introducing such a technology in a new environment (the orchestras, theatres, recording studios, etc.) for building an audio spatialization system.

ARTEC GROUP and A&G have a specific interest in distributing I'M EASY systems, as well as smaller products derived from I'M EASY parts in both national and international markets. A&G and ARTEC GROUP knows very well the market of Computer Music, A&G has a worldwide commercial network as Distributor of Computer Music Products.

I'M EASY architecture overview

A prototype of I'M EASY architecture for sound spatialization will be implemented, by using HPCN technologies relative to parallel architectures. The implementation of the I'M EASY prototype involves both hardware and software aspects.



The components of I'M EASY architecture, where the dotted line specifies the components related to the project, are:

1. DSP board: it is a multiprocessor board to be installed in a PC-compatible computer, and it is based on a special-purpose digital signal processors AD SHARC 21065L, totally dedicated to the processing of audio signals; multiple boards can be directly interconnected in various configurations.

2. Audio unit: it will be independent of the life of the product. The audio unit can support communication standards, such as ADAT, and , with a newer version, some other professional standard such as TASCAM. In this way, the product could be distributed separately even to customers that already own such a unit.

3. Microprogram module set: it is the collection of algorithms to be tailored on the SHARC assembler and board architecture, following the slot-organized architecture of the multiprocessor DSP board; the microprogram modules implement all the necessary processing functions, for both spatialization and conventional

sound processing.

4. Low-level interface: it is the set of libraries that implement an interface layer between host and DSP boards, taking care of the initialization procedure relative to the DSP boards and the audio units, and taking into account the asynchronous communication approach of the AD SHARC 21065L DSP.

5. Control software: it is the application that provides a consistent and complete environment in which it will be possible to handle the whole system, including automation capabilities, dynamic and static reconfiguration, cue list handling, external synchronization.

Note that the data exchange capability with the host PC functionality will grant the compatibility with other Pro Audio products, such as Digi-Design.

The host software set is composed of three elements, which are:

The microprogram module set

This is the heart of the whole system, implementing all the necessary algorithms to process input signals. The microprograms take advantage of the highly partitioned structure of the custom DSPs, and are hand optimized to ensure maximum performance. The main blocks have already been developed and field-tested in their basic functionality on the SPARK board. To enter the commercial market, more microprogram modules are needed, in order to offer a large set of options to the customer. Furthermore, the existing modules must be modified following the high level specifications of the I'M EASY architecture. A custom interactive programming environment is already available for the custom DSPs.

The low-level software drivers

To control the system, a library to access all the functionality of the boards will be developed. This library will depend on the specifications of the entire system. The software library currently employed has been developed mainly for testing and debugging purposes and has to be revised.

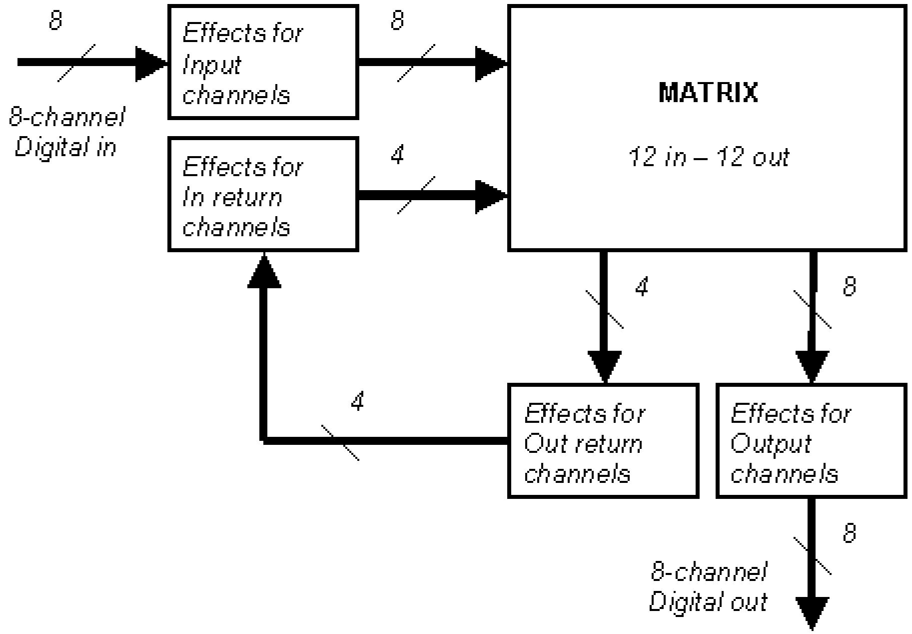
The control software

Being a reconfigurable, time-slot division multiprocessor system (multiple boards are employed, each board with 4 processors), the task of handling the current configuration of the system is not trivial. Furthermore, typical audio applications dictate the need to partially change the current configuration without introducing audible artifacts (clicks, pops, interruption of sound) into the active processing paths. This requirement strongly influences the structure of the control software layer.
Its main tasks will be: dynamic reconfiguration of parameter sets, algorithms and control structures; parameter typing; real-time specialized control of selected parameters in response to external control sources; handling of control sources (MIDI, RS-232, RS-422); handling of external synchronization protocols (MTC, SMPTE, VTC); and automatic assignment of resources (time slots, internal and external connections).

The DSP level

From a logical point of view, the DSP level architecture can be described as follows:

- 8 physical input channels
- 12 input channels with a logical input patch-bay
- a 12 X 12 gain matrix
- 12 logical audio output channels
- 8 physical output channels
- 24 effect processor, one for each input channel and one for each output channel.



The EZ-SOUND board will use these DSP resources as follows:

1. selects 12 logical audio inputs from the input patch-bay
2. processes each audio input using one effect processor
3. sends each one of the processed input signals to a "row" of the gain matrix
4. the gain matrix re-scale each input 12 times with 12 independent SW controllable weights, and accumulate the result on the 12 gain matrix output channels ("columns")
5. each one of the 12 gain matrix output is sent to an output effect processor

6. the output of each one of the 12 output effect processors is used as final output of the board.

The source of each one of the 12 audio inputs of the board will be SW configurable and will be selectable from:

- 8 -channel Digital inputs
- effect return lines
- host supplied audio signal as multimedia drivers and audio plug-ins.

The 12 audio outputs of the board will be used in this way:

- each one of the first 8 audio outputs is sent to a digital output
- up to four selectable audio outputs can be sent to an effect send line
- all the audio output can be used as input signal for host software like multimedia drivers and audio plug-ins.

The availability of host supplied input and outputs audio signals has to be considered only as an architectural specification introduced with the goal of offering an open feature for future enhancement of the EZ-SOUND software. The implementation of audio drivers and plug-ins is not part of the I'M EASY project.

Each one of the effect processor will consist of:

- an input gain
- a graphic equalizer
- a delay.

The input gain will be smoothed by the DSP chips. This means that the high level software can change it without worrying about noises ("click"). The input gain can also be negative: this results in a phase inversion of the signal.

The graphic equalizer will consist of:

- a low pass filter with variable cut-off frequency
- a high pass filter with variable cut-off frequency
- a band pass/reject filter with variable frequency, bandwidth and gain.

The delay processor is currently under project. As the board can access a large memory range, the maximum delay time will have to be scaled considering the impact of the cost of the memories on the final cost of the board.

The gain matrix will consist of 144 accumulation nodes organised into 12 rows, one for each input, and 12 columns, one for each output. Each node will have a software controllable gain. The gain is smoothed in order to avoid "clicks".

The Control Level

The control level software is a set of two libraries:

- EZSOUND library: a C++ library containing low level functions needed to access all of the board resources,
- EZSPACE library: an intermediate C++ library, based on services supplied by the previous one, which implements the spatialisation algorithms by controlling the relative gains on the matrix.

These libraries allow to interact with the upper and lower software level. The libraries will allow the higher software level to access more than one board at a time. If more than one board will be controlled from the same user interface, that is if a multi-board application will be implemented, the coherence of the behaviour of the boards is assured by the high level software.
The multi-board approach will allow adding an extra feature to the system: modularity. There are some practical situations in which 8 physical inputs and 8 physical outputs are not enough. A host computer containing more than one EZSOUND board can manage these situations by simultaneous interaction with all the boards.

The API contained in these libraries will be implemented so as to a high-level 32-bits development environment like Visual Basic 6 or C++ can easily call them.
The data formats used as parameters are those typical of 32 bit developing systems for Windows 95.

The EZSOUND library

EZSOUND low-level library manages:

- all the requirements to access the resources of a memory mapped, PCI standard and Plug'n'Play standard compliant board;
- the initialisation (boot) of the board from host, that is the firmware loading on the four DSP, and the initialisations of all the needed parameters;
- the installation and the management of a bi-directional message based communication layer between the host and one or more of the DSP processors;
- services destined at the highest software level, to access firmware resources through specific functions of the particular DSP application.

EZSOUND library has been implemented as a 32 bit Windows dynamic link library EZSOUND.DLL containing:

- the management of the PCI Plug n' Play interface to the board,
- the "multi-board" logic, that is the management of a spatialisation session, which uses more than one board,
- all the functions needed to initialise and control the board resources,
- all the functions needed to interact with the DSP chips,
- the interface (API) to the upper level.

The API will supply functions to allow the host computer to:

- initialise a board,
- manage the input channel assignment from the input patch-bay,
- control the 24 effect processors parameters,
- control the 144 gains of the gain matrix,

- send one or more host generated audio signal to the processing section by using a virtual player,
- receive one or more audio signal coming out from the processing section by using a virtual recorder.

The board access functions provide for multi-board applications, i.e. they allow access to many boards at the same time.

The EZSPACE library

EZSPACE libraries implement the spatialisation mechanism. It will be initialised by the high level program with all the information needed on the actual loudspeaker set configuration, including the Cartesian co-ordinates of the loudspeakers. Then the high level software will supply, at runtime, the position to all the sound sources. The position of both the loudspeakers and sound source is expressed as three-dimensional co-ordinates. The library use all these information for computing the relative gain of a source through a loudspeaker.
As there is no commonly accepted "good" spatialisation algorithm, this library has been separated from the EZSOUND, the low level library. In this way, it is simple to develop and test different algorithms. Moreover, updating a single library do not effect the other.

The EZSPACE.DLL library manages the computing and gain setting for all matrix nodes and offers a set of higher level functions for sound spatialisation.

Different spatialisation algorithms are under testing, to investigate the possibility of allowing the user to choose between different ones.
The spasialisation mechanism will use the following input data:

1. The description of the loudspeaker configuration consisting of the number of loudspeakers. For each loudspeaker the following data are provided:
    - The loudspeaker position (x, y, z);
    - The position of a point (different from the loudspeaker position) laying on the irradiation axis of the loudspeaker. This is needed to calculate the loudspeaker orientation;
    - An attenuation graph expressed as function of the irradiation angle;
    - An attenuation graph expressed as function of the distance from the loudspeaker.
2. The position (x, y, z) of each sound source.

    The information of the loudspeaker configuration will be supplied by the high level software, once for each performance. The position of the sources will be updated at run time when the position changes. There is no need for a very high update rate, as the smoothing mechanism included in the DSP level will interpolate between different positions in time.

While a detailed study about the best set of available spatialisation algorithms is expected, a very simple one has been implemented in order to test the software architecture. It only uses:

1. the Cartesian co-ordinates of the loudspeaker
2. the Cartesian co-ordinates of the sound sources
3. the distance factor of the loudspeaker, which is different for each loudspeaker. It specifies how fast the gain will decrease if the distance between sound source and loudspeaker increases.

The algorithm consists of calculating the square of the distance between the sound source and the loudspeaker, and raising the distance factor to the distance power. As the distance factor is less than 1, the gain will decrease while the distance increases.
This is not a complicated algorithm as it does not take care of loudspeaker directionality and does not use a correct mathematical model of the physical situation, but it is

very simple and does not use much CPU time. A test of this algorithm, running on the SPARK board, has given acceptable results, so it is one of the candidates for the final application.

High Level Control Application

The I'M EASY high level control application (HLCA) is the visible part of the spatialization system and it is the main interaction component. Among others, the main features are:

- Definition of a virtual 3D room or stage as the base of the spatialization system
- Full interaction in the 3D environment: Top view, Left view and 3D view
- Possibility to move and drag elements in the space, to draw directly in the space the input source trajectories and to move interactively an input source without trajectory (hence spatializing the sound of that source) while other input sources are in running mode
- Editing of the single trajectory step (position, timecode)
- Hierarchy-ordered elements to split session in more usable parts
- Handling a session, scenes, inputs and outputs
- Saving and restoring all the user performed spatialization works in IMEASY document format
- Defining special sound effects for the input that the user has chosen
- Comparing and eventually reorganizing the timing of trajectories via a time scheduler
- Playing mode to activate the spatialization board

(Click on the image to know more about the functionalties of I'M EASY related to the corresponding screenshot)

Main Window

Input source positioning

Drawing a sound trajectory

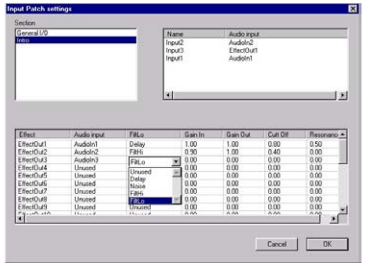
Another view of the sound trajectory

Full views of the trajectory

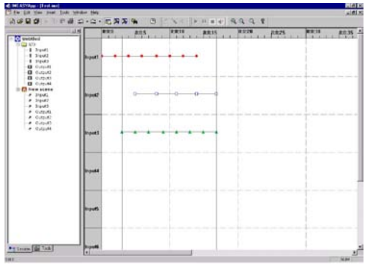
Drawing trajectories with different views

Trajectories defined

Source properties



Input patch setting



Time scheduler

Playing mode

The HLCA will be installed by the I'M EASY user, with the help of a standard installation set of media, manuals and the EZ-SOUND board. The installation will be modular, meaning that user will be able to tailor the high-level control application according to his needs.First, the user will install the audio spatialization board in one of the available slots of the I'M EASY workstation. Next, the user will run the installation utility to copy the high-level control application on the I'M EASY workstation hard disk.

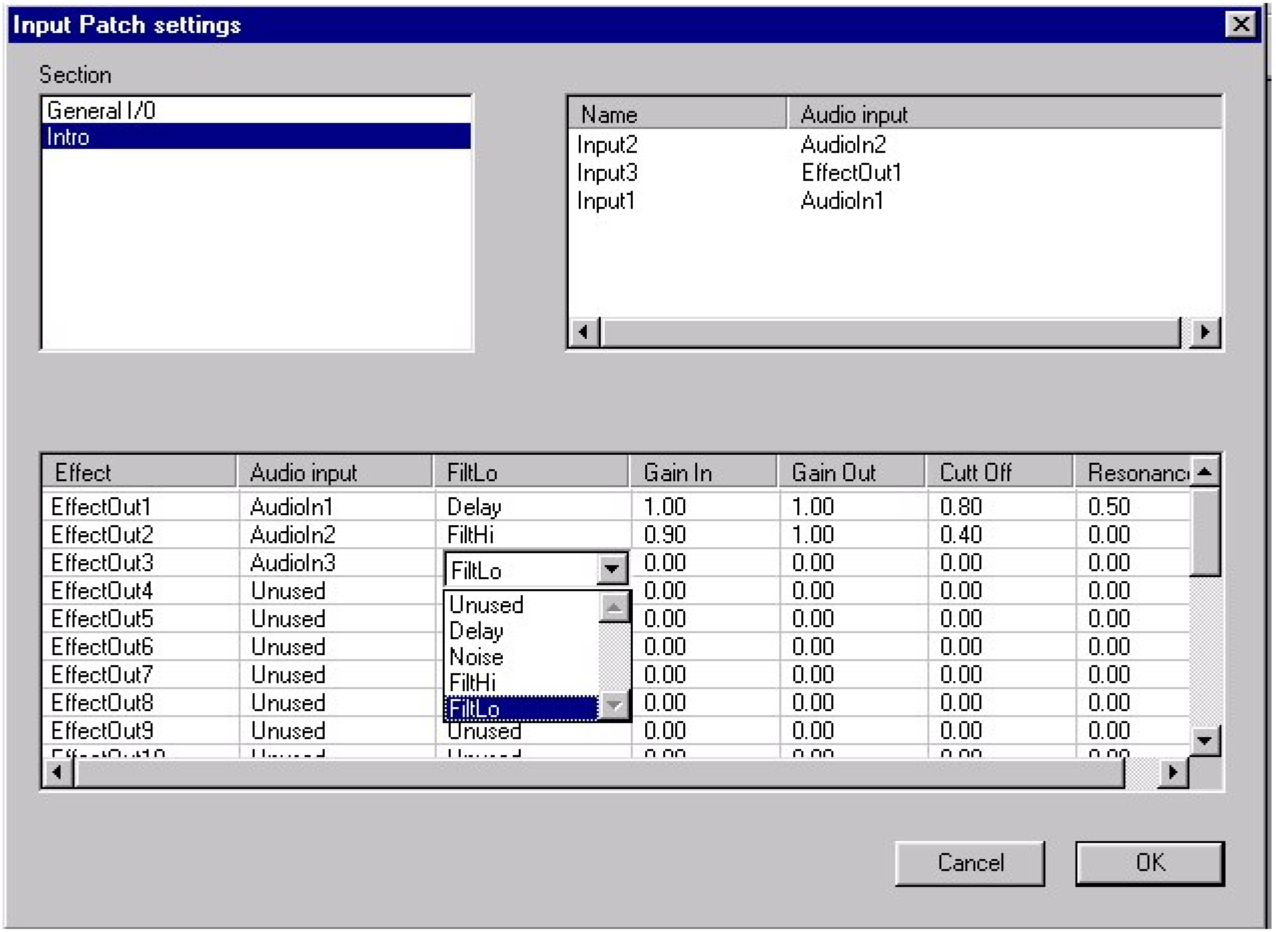
HLCA requirements

The requirements of the high-level control application at the user inteface level are:

- Sound source handling
- In and out source testing
- Outputs mastering
- Mixing
- Session handling (workspace and sessions)
- Scenes handling
- 2D drawing of the stage map
- Creation of a sound source path over the stage
- 3D stage maps handling

Input patch setting

It is possible to define special sound effects for the input the user has choosen. Each virtual input can be associated to an Audio input (from board or from file), as well as to an Effect generated from an Audio input.

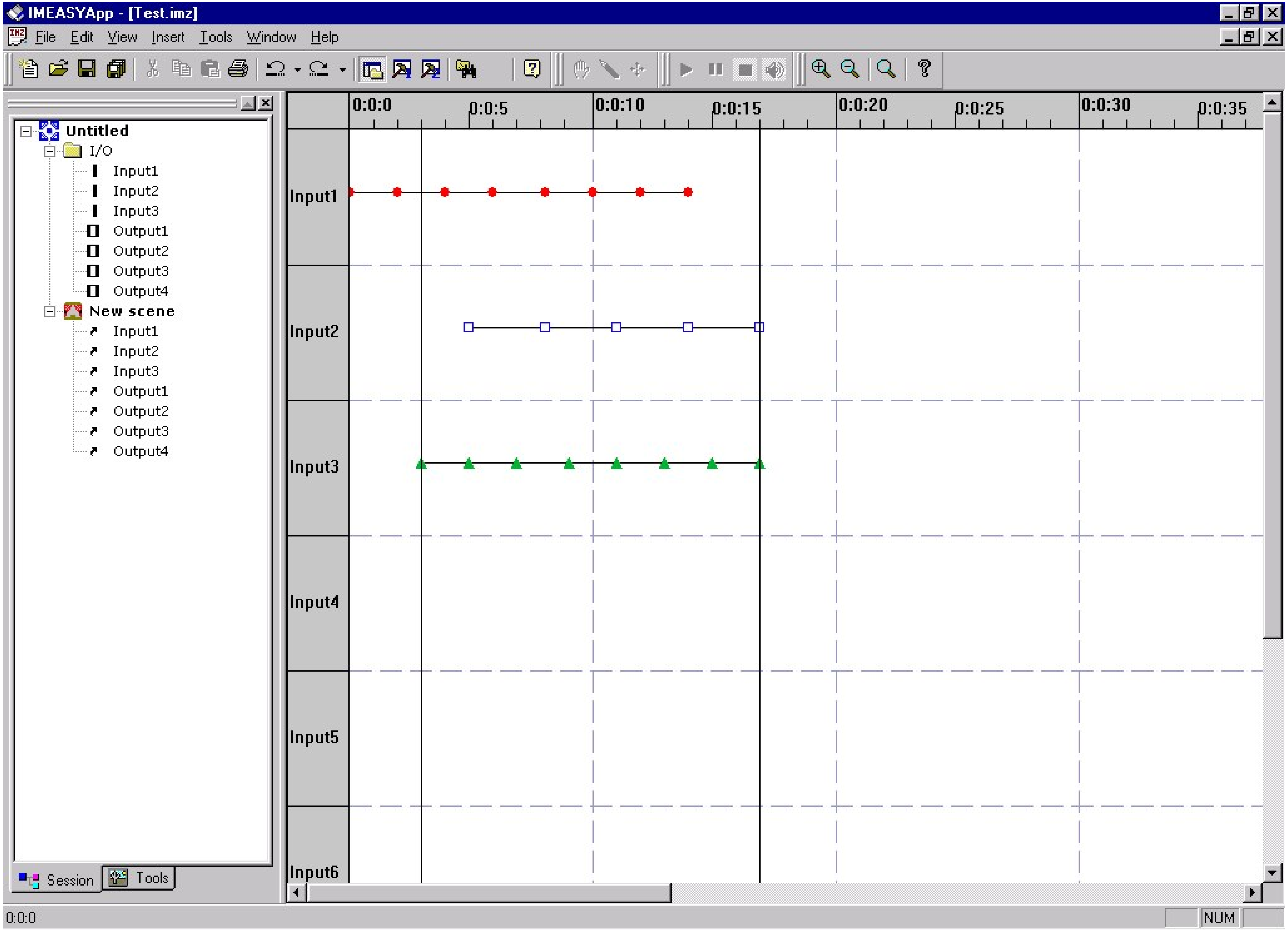


Time scheduler

When all the trajectories are inserted in the scene, the user may need to see the planned scene from the time point of view. A tool that makes a temporal snapshot of the session and permits the user to compare and eventually reorganise the trajectories seems to be really useful.
The Time Scheduler is a window located on the "views" field of the Main Window. It can share the "views" filed with the 3D scene, but when it is selected, the system changes the control set and shows the zoom palette. On the top of the Time Scheduler, the temporal axis is present and its scale changes together with the zoom level selected.
For each input, the task of this tool is to show the temporal distribution of the trajectories nodes and to move the trajectories along the time. When a trajectory is selected (to select a trajectory click with the mouse on the trajectory) at the beginning and at the end of the trajectory, two lines are drawn. These lines show when the trajectory begins and ends. They are useful to compare the selected trajectory with the others or, when the user moves it, the lines show immediately the new time position.

In this example, three inputs are reported in the Time Scheduler. The "Input 3" is selected and it is clear that its trajectory begins after three seconds that the session is started and terminates, together with the "Input 2" trajectory, at seventeen.

Screen 11


IMEASYApp - [Test.imz]
File Edit View Insert Tools Window Help
Untitled
I/O
Input1
Input2
Input3
Output1
Output2
Output3
Output4
New scene
Input1
Input2
Input3
Output1
Output2
Output3
Output4
Session
Tools
0:0:0
0:0:5
0:0:10
0:0:15
0:0:20
0:0:25
0:0:30
0:0:35
Input1
Input2
Input3
Input4
Input5
Input6
0:0:0
NUM