The TAU2 polyphonic music terminal: the project, its realisation and role in computer music

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Preface

In the early months of 2003 two related events occurred: the installation of the audio terminal TAU2 at the Museum of Computing Instruments in Pisa and the inauguration of a centre for the restoration of audio objects at the L. Cherubini Conservatory of Music in Florence, where the now obsolete TAU2 had been for many years. The restoration of tape recordings, LP records, and other material produced on the TAU2 system and its control language TAUMUS is one of the first activities at the centre (MART for Music, Audio, Recovery/Restoration, Technology).

These events and the fact that I had participated directly in the first projects involving the use of computers in music have made me reflect on what has happened since the 70's in the development of computer music in Italy. I readily accepted the invitation to review the activities in Pisa in this period, in particular the unforgettable experience involving the creation of the computer music system TAU2 – TAUMUS. Prof. Franco Denoth of IEI (Istituto Elaborazione Informazioni) at the CNR in Pisa proposed the project to meet the needs of M. Pietro Grossi's initiatives in the field of Computer Music. Thanks to the partnership between experts at the CNUCE Institute and hardware designers at IEI, coordinated by Prof. Denoth, the system was functioning by 1975.

The ambitious goal was the construction of an audio terminal to interface like any peripheral device to the mainframe IBM 360/67 of the CNUCE, to allow the electronic production of sounds and the execution of polyphonic and polytimbric music in real time using the program TAUMUS. Thanks to the creative use of additive synthesis and other specific solutions, the TAU2 produced good phonic quality and allowed direct interaction between the user and the computer. Entrusting the execution of the pieces to a system other than the one used for elaboration meant significantly lower usage in the CPU time of the computer with respect to other systems of the period.

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Since we were dealing with an application in a new area without definite standards, we faced significant difficulties, gradually overcome in time, in the definition of specifics, design, and realization of the equipment. On the other hand, at that time there were only a few isolated initiatives in a few centres abroad and in Italy, whose work was little known. There were no specialized publications with information on the activities of the various groups. For example, the «Computer Music Journal» began to be printed at MIT (Massachusetts Institute of Technology of Boston) only in 1977 and the first issues arrived in Italy when the TAU2 had already been built and functional for some years.

The definition and choice of how to use the system, the production of the pieces, interactions with the musicians, teaching and demonstrations with TAU2-TAUMUS, to which I refer in this article, have combined to create an original musical environment that contributed notably to the development of musical informatics in Italy. At any rate, I think it would also be useful to say something beforehand about the experiments developed to use the new 'electronic instrumentation' in music, citing also the state of the art and the problems encountered in the use of computers to synthesize sounds.

Introduction. From electronic music to computer music

The close relationship that exists between music, mathematics and technology (the mechanics of musical instruments), recognized at a conceptual level from the Greeks on and successively developed over time is an accepted aspect of the general field. One need only think of the enrichment of the typology and the characteristics of traditional musical instrumentation that has taken place over the centuries. This connection has become even closer in the 20th century with the use of electronics and innovative practice in contemporary music.

Undoubtedly, electronic technology has resulted in notable advantages for the spread of music among the educated, almost exclusively the prerogative of the elite for centuries, and the more popular and experimental, forcing it to emerge from its restricted spaces and localised environments. The use of amplifiers, microphones, speakers, disks, tape players and radio has permitted the enjoyment of all kinds of music in any place and at any moment, while not diminishing the suggestiveness of live concerts, but more redefining their function as a means of spreading new musical messages.

The notable qualitative jump came with the use of electronics to not only manipulate signals taken from various sources, but also generate sound *directly*. The connection of equipment that pilots the speakers enriched with *oscillators* was developed and used as new «electronic musical instrumentation», capable both of imitating the sounds of traditional musical instruments, and to produce new and more extended sound texture. An historical overview of electronics in music can be retraced at <http://www.obsolete.com/120_years/>.

This was all accompanied by the tendency of more eclectic composers of modern works (20's to the 60's), intent on going beyond classic rules of harmony and composition, to emphasize the timbre characteristics of sounds, the thicker density of the

Birth and evolution of the informatics sector at Pisa: IEI and CNUCE of CNR

Immediately after the development of the first digital computers in the United States, the CEP (Calcolatrice Elettronica Pisana – Pisan Electronic Computer), the first large Italian digital computer, was constructed at Pisa in the second half of the 1950s. It is now on exhibit in the Museum of Computing Instruments in the same room with the TAU2. The project and its success were due to a group of researchers and technicians from the Centro Studi Calcolatrici Elettroniche (CSCE) formed thanks to various scientific partnerships and contributions, principally among which were Olivetti SpA, University of Pisa, and other agencies (Denoth, 1991).

The CEP began to work in 1960 and was officially presented in 1961. It was developed in part with thermoionic tubes and in part with transistors and can be placed between the 1st and 2nd generation of computers. It functioned locally only, had one principal console (single-user), and for I/O units had some card punchers and paper tapes photo-readers.

At that time, in 1962, just graduated in radio technology, I began working at CSCE when the Centre passed to CNR. As a first job I participated in the expansion of CEP which consisted of the addition of an external memory, a bank of 8 magnetic tape units (the 'large reels' of the Ampex), interfaced with the machine by means of a complicated control device, a 2m x 2m 'closet' full of electronic transistor boards.

The CEP was operational for six or seven years and the activity related to its planning and programming contributed to the growth of the local Pisan and national scientific community in this developing sector of informatics. In 1968 the CSCE obtained an important recognition in becoming an institute of CNR – the Institute of the Elaboration of Information (IEI-CNR).

At the same time the University decided the Pisan environment was appropriate for another important centre, with the principal goal of supporting the growing needs of computing in the Italian scientific community. CNUCE (University Centre for Electronic Computing) was created in 1964 and IBM machines were installed, beginning with the 7090 and followed by mainframe multi-user systems 360/67, 370/168, etc., working in time-sharing.

[See image at p. 100: Modular components of the CEP and frames showing parts using thermoionic valves, built at CSCE]

notes (beyond the predefined framework of the pentagram) and introduce into music concepts like «sound processes in evolution». In that sense, the new rules of composition could exploit mathematical relationships, laws of probabilistic distribution and various algorithms, whose execution was difficult with the instruments available at the time. In the 50's the first laboratories of phonology were founded, based on the use of analogical instruments like oscillators, ring modulators, filters, effects, etc[1]. All of these modules had inputs and outputs on panels: by wire cabling various routes (patches) could be realized and using manual controls you could obtain the desired types of sound and register on tape and manipulate afterwards by 'cutting & pasting'. With this instrumentation and the new composition rules various works and pieces of music were produced and identified with the term 'electronic music' and 'tape music'.

[1] Laboratories were started in Europe in Paris, Cologne and Darmstadt, in New York in the US, in Italy at the RAI studio in Milan, in Florence (the S2F M founded by Pietro Grossi at the Conservatory Cherubini)

In the 50's the first electronic digital computers appeared. They were costly and large, constructed initially for scientific and military reasons. Shortly they began to be used for many other industrial and civil, even artistic applications. One of the first notable applications of electronic digital techniques was the modernization of telephone switchboards and later the numeric coding of vocal signals. The results of these achievements have had an important impact in many other related fields.

Some far-sighted pioneers intuited the possibilities of using computers in music and during the 60's began research and experimentation in analysis, archiving, and composition of musical texts, analysis and synthesis of sounds, etc. The most evident innovation was that the greater part of the work and research was carried out by software procedures; that is, *programming* that could be executed by computers. From the beginning this new activity was identified in its totality by the term «Computer Music» (Tenney 1969).

In Italy interest in the use of computers in music developed in the 70's with the birth of personal initiatives and study centres, each with its own calling and specialisation. The new working methods were experimented on and shared by Maestro Pietro Grossi. Already by the middle 60's the objectives, posited and presented in seminal articles (Grossi 1969), then found more efficient means of completion at Pisa, where the first important scientific centres for computing had been formed, CSCE and CNUCE, subsequently institutes of CNR.

A network of connections, based on the SIP telephone company data network, was developed. This allowed remote connection (via modem, switched or dedicated phone lines, etc.) and the use of adequate calculation resources at different locations. A complex structure organized the execution of various duties, with different fares based on the priority of service to public organizations, industry, etc.

The knowledge necessary for managing computers, numeric system architecture, programming languages and techniques made the introduction of a new academic discipline in the Italian university system imperative and so, in 1969, the first Degree in Information Science was instituted at the University of Pisa with a significant contribution from IEI.

Computers in music field

Use of computers for sound synthesis; the utility of an audio terminal

Already at the end of the 60's research on the use of the computer in music was very promising and suggested interesting applications. The possible use of computers in music derives from the fact that a piece can be represented and codified by a sequence of numbers both at a symbolic level (the score, the sections, and the notes on the pentagram), and (though with a certain approximation) at the level of sound associated with notes played on each instrument with the appropriate physical parameters[2].

[2] A musical note is characterised by pitch, duration and intensity. When the note is executed by a musical instrument the complex sound produced also has an important timbre attribute. The first three

At the beginning techniques and software procedures to analyze the structure of a piece and extract information about the rules of composition that characterized it were devised both to transform a piece and also compose new pieces (Tempelaars, 1972).

Soon the need to 'execute' in an automatic fashion the music elaborated or composed by the computer became evident, freeing the performer from the difficulty of execution linked to the manual nature of the instruments. By 'execute' we mean *to generate signals with the right wave forms (synthesis)* able to produce sounds with musical attributes specified by the user via loudspeaker. We know this operation is difficult, depending on the complexity of the sounds we want to produce. Even now there is no standard, efficient, precise, and definitive method for the analysis and synthesis of any musical signal. Over time various techniques have developed based on different approximate models of mathematical representation of signals that permit us to come close to an excellent result. These techniques have advantages and disadvantages and their choice depends on the aims of the application and the technology available (De Poli, 1983).

Two general methods were identified to elaborate a piece and perform it automatically and electronically.

a) pure digital method: beyond elaborating the structure of the musical piece, the computer is used to identify the sequence of numbers (the samples) that, applied to a digital/analog converter (D/A), allow us to obtain the analog form of the desired signal. With this method we put a heavy duty on the computer and, in most cases, it was impossible to reach a sufficient calculation speed of the samples to guarantee the correct functioning of the D/A[3]. In order to avoid this problem we had to proceed in two steps: first, we processed the score and accumulated the samples in the memory, usually taking more time than the duration of the piece itself, then the samples were read again and applied to the D/A converter at the correct speed, thereby obtaining the correct sound. The chief drawback of this procedure was that if you wanted to modify a piece, the previous process had to be repeated several times. This procedure was defined as *performance in differed time* (Mathews, 1969).

b) hybrid method: the computer processes the musical structures, produces data and parameters of the sounds, codified in an appropriate manner (at a *macro*

parameters are associated to the physical attributes of fundamental frequency, sound duration, and amplitude. Timbre depends on the wave shape or, in other words, on the sound partials, that dynamically evolve over time. In addition, timbre is connected to auditory perceptual features. However, it can be specified with proper numerical parameters even if with a certain difficulty and in an approximate way (*Il Suono*, by P. Righini, Tamburini, Milan 1974).

[3] Theoretically, the number of generated samples/sec must be in theory (sampling theorem) at least twice the signal band (two or more samples/period at the highest frequency). In practice, in musical applications, at the time considered of good quality (band of 15 KHz) and with the available conversion and filtering equipment, at least 5-6 samples/period were required. Therefore a continuous flow of at least 75000 samples/sec was needed at the D/A input. However the computers could not then cope with the requirements, considering also pre-processing and other computations for complex scores.

level) and sends them to a peripheral, a specialized synthesizer, which generates the signals (at a *micro level*) based on the commands it receives. In general this method meant less flexibility in defining the wave form, delegated to the external synthesizer, but greatly reduced the processing time. This way we were able to achieve sounds in *real time* even from the computers of the time. It's also worth point out that less *machine time* meant savings on the cost of computer use, which was fairly high at the time.

To summarise the two situations: in the first case the computer functioned as *composer's helper/conductor/instrument*; in the second case it performed the first two functions, while relegating that of *instrument* to the external musical terminal. Moreover, in order to have the advantages of the hybrid method, we had to either *readapt* traditional synthesis modules, originally manually controlled, with the right digital control (Zinovief, 1969) or *design them ex novo* (as in the case of TAU2). The expense of both solutions turned out to be directly proportional to the complexity of synthesis and control techniques adopted.

Grossi used another less elaborate solution in his first experiments. This consisted of using a digital signal (ranging from zero level to tension level) taken directly from a bit of the computer's arithmetic unit. When the computer executed the right procedures using the DCMP (Digital Computer Music Program), it produced square waves with a basic frequency variable in audio band, such that it generated single voice melodies following traditional canons or other rules of composition (Grossi, 1974).

The project for an audio terminal for working in real time

Research at CNUCE

When he sounded out Pisa (1969), Maestro Grossi had already done experiments on the electronic generation of sound using oscillators, modulators, and other analogue equipment, having proposed them for use in teaching at the Cherubini Conservatory in Florence, where incidentally he had been teaching cello for many years. Convinced of the great potential of the new numerical electronic computers in all fields of music, he turned to the institutions where he could obtain the necessary means and skills to help realise his intuitions.

From the very first experiments, he adhered to three fundamental principles: *real time* (even compromising on the sound quality), *interactivity* (local and remote), *automated processes* (including decisional ones) (Grossi, 1971).

[See image at p. 103: Machine room of CNUCE (c. 1978)]

Prof. Guido Torrigiani, director of CNUCE, and Prof. Gianfranco Capriz, director of IEI, both mathematicians and music lovers, were convinced of the utility of computers in this field and responded positively to Grossi's proposals. With the additional help of experts at the IBM Scientific Centre in Pisa, which had grown up near CNUCE, Grossi began to experiment with generating music in both differed time, with the IBM

The TAU1

In the TAU1 the original sinusoidal signals were created using square wave VCOs (Voltage Controlled Oscillator). Partitioners and analogue adaptive filters followed. Amplitude control was obtained with digitally controlled VCAs (Voltage Controlled Amplifier). The circuits were created in part with analogue technology and in part with DTL (Diode Transistor Logic) integrated circuits. A laboratory prototype comprising one part analogue circuitry for synthesising 4-harmonic two voices and one digital part, which could supply the note frequency and duration parameters in a dynamic way, was built.

[See image at p. 104: The TAU1]

I designed the digital part (Bertini, 1974), which managed the buffers to contain the sound parameters and the photoelectric reader (from those discarded by CEP) that extracted data with information on the piece to perform from paper tape prepared previously by Grossi's collaborators at CNUCE.

Although provisional, this was still a way of working in differed time. In addition, because of the noise emitted by the mechanical tape reader, you had to listen to the piece after having recorded the sounds on a magnetic analogue tape. Some of the pieces created with TAU1 have been included on the demo LP, "Computer Music", Musicali Fonos (1972), which also contains pieces created with DCMP and others with the IBM 1800 system and S/7 with the D/A converter.

1800 and S/7 systems equipped with a D/A converter, and real time, with the DCMP procedures for direct output of the square wave signal. In addition, he began to use the CNUCE's data network for remote experiments. In all cases, the noticeable amount of computing time required by the computers constituted a major inconvenience, thus IEI was asked to participate in finding a solution to the problem.

Research at IEI: the creation of TAU1

At IEI Prof. F. Denoth responded to the invitation to take on the problems relating to computerized sound synthesis, having developed a certain expertise in digital-analog technology in the area of avant-garde research in the biomedical sector[4].

The hybrid method was pursued, with the decision to design and realise a special audio, polyphonic and polytimbric terminal that could be connected to IBM computers like any peripheral and possibly not interfering with normal work of the time-sharing systems.

The first step was to evaluate and experiment certain solutions for adopting synthesis techniques. The work was initially developed in a physics thesis (Chimenti-Denoth, 1976) and subsequently expanded with financing split between other research groups within the CNR research project entitled "Analysis and Synthesis of the Voice and Sounds."

[4] Following the successful realisation of an original *self-synchronising cardiac pacemaker* (patented) and in order to better contribute to important research on the use of numerical techniques in the analysis of elettromiografic and heart signals, IEI founded a research section in 1967-68 under the direction of Franco Denoth,. Its purpose was the application of informatics in the bio-medical field.

Among the various techniques possible, Denoth chose additive synthesis with sinusoidal components, a courageous choice at the time because it was difficult to implement (and, in general, still difficult until recently even with modern DSPs). In this way, TAU1, the first laboratory prototype, was created.

Design, architecture, and operating criteria for the TAU2

The experience of the TAU1 was useful in evaluating the definition of the quantisation intervals and the validity range of the parameters. For instance, it was possible to dynamically change frequencies (so obtaining glissando), amplitude control of the pitches, the minimum durations to assign to the metronome and to evaluate their effects. Timbres were pre-fixed using manual switches.

At this point we began to understand the complexity of the problems inherent in computer music and its multidisciplinary nature, which involved: electronics, signal processing, computer programming, informatics, and obviously, music (in many of its facets), in addition to electro acoustics, psychoacoustics, etc. The latter discipline studies the features of auditory perception to evaluate the differential thresholds with respect to the variation of physical quantities (frequency, loudness, duration and timbre of sounds), which play a role in determining the quantisation levels of the associated parameters. In fact, the aim is the simulation of the continuous variations of such parameters, i.e. the way they work in the real world. The number of definition levels of the numerical parameters was kept as low as possible in order to allow the computer manage a small amount of data.

Besides the above considerations and the experiments already performed on the TAU1, the definition of the design features of the TAU2 required to take into account different requirements, of which the main ones are:

– Acoustic-musical requirements: the possibility of producing a sufficient number of notes on different channels, so that pieces with a certain complex structure could be performed, without the use of 'play-back'
– Functional requirements: to work in real time by interfacing to a time-sharing system with minimum computing time for the system
– Implementation requirements for the terminal: realisation with reasonable size, construction time and cost
– Computer programming facilities requirement: intuitive interface with the system that allows a user-friendly working environment

The main problem was to find a non-trivial configuration of the quantitative design data of the terminal to assure the best balance of all these requirements. After having evaluated different solutions for the techniques to adopt for signal generation and having tested new components for the TAU2 realisation (since the solutions adopted for the TAU1 had not been completely satisfactory), the range of variability of the parameters was defined and, in general, all the other design and realisation requirements for the audio terminal (Bertini-Denoth, 1975).

The organisation of the system is shown in Fig. 1 [see p. 104: Architecture and system resources TAU2-TAUMUS]. In short, note the following:

– TAU2, made up of two physically distinct parts:

a) digital unit – receives data through the connection with the IBM system, memorises it in a circular buffer, interprets it and supplies it in the right format and time to the analogue part.

b) audio unit – generates elementary signals, modulates them in amplitude and combines them to produce complex musical signals, based on the parameters present at its inputs.

– IBM system with all of its hardware and software components, which hosts the TAUMUS programme, the archive of musical pieces, and the procedures for elaborating parameters and forwarding data to the connection with TAU2

– terminal (telewriter+video) next to TAU2, by means of which the user dialogues with the IBM computer, activates TAUMUS, inputs data and programmes, carries out the commands and orders the execution on TAU2.

Operational principles and musical characteristics of TAU2

Audio signals are, in principle, obtained with a variable spectrum harmonic additive synthesis process, programmable at 'short time intervals' (in the order of a few ms. Details on this synthesis model are described in the appendix). The starting sinusoidal signals have fixed amplitude: the lowest frequency signal, the most fundamental, identifies a 'voice' (it can be a tone of the equal tempered scale or any intermediate sound taken from the available sounds), while the evolution of the amplitude of the associated harmonics, that can be adjusted dynamically for each harmonic using dedicated VCA's (Voltage Controlled Amplifier) allows us to shape the timbre of the sound.

TAU2 contains a set of wave generators with a range of 324 sinusoidal signals at very stable fixed frequencies (implementation details are given below) that range from 32,7 Hz to 16.425,1 Hz, with an interval ratio 1/3rd of half-tone, fine enough to simulate an almost continuous scale.

The audio unit incorporates three identical channels. In Fig. 2 the main components of a channel are shown. There is a unique generator bank that supplies the sinusoidal signals at a continuous rate to all the three channels. While keeping in input the numerical code F_j that identifies a voice ($1 \leq F_j \leq 255$), a special hardware structure, made of thousands of digitally controlled integrated analogue switches (selection and mixing network), selects in the generator bank and directs the corresponding fundamental frequency $f_0(F_j)$ to the first exit (B1 bar). Automatically the harmonics $2f_0(F_j)\ldots7f_0(F_j)$, associated with that fundamental, are also sent to the other six exits B2 …B7.

[See Fig. 2 at p. 107: Structure of an audio channel. Frequency selection and mixing network, B1-B7 harmonics summing bars, Mi =VCA harmonic amplitude and output intensity level control circuits, K = Selectors]

The selection network can manage up to four distinct voices at the same time and it selects the related harmonics from the bank for each by directing the signals to the respective bars. Therefore in any bar B_n there is a signal made of the sum of up to four harmonics. The bar output signals are amplitude adjusted before entering the channel summer. Eight dynamic levels are possible (27 dB dynamic range) to assign the channel timbre (M_1 .. M_7 blocks). The output signal of a channel can be volume controlled with 16 possible levels (29 dB dynamic range, M_I block). In this way the four voices have the same timbre; that is, the same amplitude ratios between the harmonics of the same order: this means four instruments of the same family.

The parameters that control sound can be chosen for each channel and are composed of frequency codes $F_{j1}\ldots F_{j4}$ that identify the voices to create. Other sets of values $A_1\ldots.A_n$ are used for the harmonic amplitudes, I (intensity) for the channel volume and D for time duration.

The execution sound duration (t in ms), corresponding to a given set of fixed parameters, is defined by the following relation:

dt = D×T , where:

D = is a parameter that can be set by software by TAUMUS and can take integral values from 1 to 31;
T = base time (or metronome time) is pre-set in the hardware on the TAU2 and can be manually adjusted between 1 ms and 999 ms; it is normally set at 10 ms ($1/100^{th}$ of a sec.).

The control panel allows testing of different timbres by manually adjusting the Ai and I levels for each channel. This procedure can be useful for setting the proper levels or for other working tests.

The U(t) output signal of a channel can be represented by the following expression:

$$(6)\quad U(t) = g(I) \sum_{n=1}^{7} g(A_n) \sum_{j=1}^{4} A_0 \sin(2\pi n f(F_j)\, t + \varphi)$$

dove:

g = funzioni di trasferimento dei modulatori
A_0 = ampiezza dei segnali di ingresso
$f(F_j)$ = frequenza dei segnali di ingresso
φ = fase dei segnali di ingresso.

Therefore U(t) is a periodic signal defined, within each interval (dt=D×T), by the parameters F_j, A_i and I. It has an extended dynamic range (from a few millivolts to a few volts). The signal phase φ is random, i.e. non programmable. This condition caused problems, with respect to the expected sound results. Hardware filtering is performed at the beginning of each definition interval in order to reduce the *clikcs* when the parameters are changed. In practice, special attention must be paid to the choice of the parameters values, to avoid amplitude variations in the output signals that are too large. For instance, a gradual scaling of the levels can be used. A signal example obtained with the TAU2 is shown later.

Organisation of data

The data needed to identify and/or direct the parameters to the proper channels are attached to the parameters when sent to the TAU2 from the computer. A protocol was therefore defined that included special sets called «musical instructions» with variable format (a kind of primitive MIDI protocol). They are shown in Fig. 3; the main instructions are: a) Timbre instruction, that contains the A_i parameters related to the three channels, b) Sound instruction, that collects the frequency values and other bit fields for controlling the circuits for special effects (vibrato, tremolo, reverberation) attached at the output of each audio channel. The first byte contains the operation code (CO), and the field IC (channel address) specifies the channel to activate. MT means Timbre Module (with the A_i amplitudes) and MS means Sound Module (with the four F_k frequencies, Intensity and ES special effects) for each channel. The R bits are reserved for future use

[See Fig. 3 at p. 109: Instruction Format: a) Timbre; b) Sound; c) a + b; d) Invariable Parameters; e) End of Piece].

All types of instructions contain the duration parameter, with the exception of the End of Piece instruction. It is possible to speed up or slow down the execution time of the piece using D through the TAUMUS program, or manually operating the TAU2 control panel, for fine adjustments.

The spectrum (and so the wave shape) can be adjusted at the channel level. The different spectral combinations are 2^{21} per channel and allow a large timbre range that can be varied in sequencer, anyhow, within the same piece. This feature permits a continuous modulation of timbre (morphing).

There are 12 voices available at the same time, that are divided into three distinct timbres, for a total of:

4 voices x 7 harmonics x 3 channels = 84 output signals from TAU2; stereophony plus the central channel, make it possible to realise an efficient sound spatialization.

Functional description of TAU2

The functional block diagram of the main TAU2 units is shown in Fig. 4 [see p. 111]. A few secondary components are not shown, such as the stabilised power suppliers, the amplifiers, the mixers etc.

TAU2 is connected to the host system that operates in time-sharing, by means of a direct parallel cable, with 16 bit of information and some other control signals. The transmission proceeds at about 50 K double byte/sec, in blocks of 1024 double bytes. The digital unit includes the input interfaces for the electrical adapters and for synchronisation of the transmission control signals. The data blocks that contain the musical instructions are memorised *per byte* in a cache memory (4 K byte buffer), using special operations managed by the control unit. During normal operation the control unit extracts the data byte by byte from memory, interprets them and assembles the musical instructions in the right format into intermediate registers (Reg. Sound and Timbre Instructions). As soon as the time associated with the current sounds to produce for a certain duration D has expired, the control transfers the new parameters in parallel from the Instruction Reg. to the Output Reg., substituting the expired ones. This process is replicated up to the end of the execution of the piece.

[See Fig. 4. at p. 111: TAU2 structure: A) Digital unit: IE-Input Interface, II-Internal interface, MT-Cache Memory UC-Control, RI-Instruction Register, RU-Parameter output register, PC-Control panels, O-Metronome clock B) Audio unit: C1, 2, 3-Audio channels, ES-Special effects, TM-Manual keyboard]

Control is based on a microprogrammed structure in order to cope with the type of functions to be executed and the requested computation speeds. This allows a better organisation of the project and easier maintenance, with respect to other solutions. The control functions are memorised in a special ROM (Read Only Memory) made of discrete components and implemented with a 'horizontal' kind of programming; therefore some operational components (networks and e registers) work together with a certain amount of parallelism (Maestrini, 1972). The memory reading procedure is executed in background whilst the data writing is executed upon interrupt to respect the connection constraints. The passing of the new parameters occurs at the moment the 'time expired' event for the active instruction.

Memory is divided in two blocks: while in the first the instruction bytes of the instructions to be assembled and substituted to the actual ones are read. In the other it will be possible to insert the bytes for the blocks of incoming instructions for the next part of the piece to execute and so on. The memory read/write address counters are realised at unit increment and organised to make memory work as 'circular buffer' (an operating way similar to the modern DSP microprocessor applications). A limited memory capacity was enough (4 Kbyte) to compensate for both the channel response time delays to the TAU2 requests (sent when there is an empty block), and the different arrival speeds of the input-output data from the digital unit.

[See image at p. 111: One of the two ROM cards, diode read-only memory]

When the user sends a command from the video terminal to process and execute a piece, TAU2 is active in 'stand by', and ready to receive the first block from the channel, sent automatically by TAUMUS. Afterwards, the control unit manages transmission, sending the request for new data blocks, as soon as half buffer becomes free. Other details in (Bertini, 1978).

Past working experience had taught that real time, without sound interruption caused by 'no data available', was almost always assured. The exception was during the central hours of the days when over one hundred users were connected to the time-sharing system or in case of overload in the specific channel to which TAU2 was attached.

As far as the audio unit is concerned we already pointed out in detail the channel structure. This structure was identically replicated for the three channels. Later on, a new circuitry with mixed analogue/digital technology was added for the special effects (reverberation, vibrato, tremolo, chorus). The latter can be manually or software controlled, and in the latter case using special bits of the sound instruction. Along the signal path at the output of each channel other analogue manipulation circuits were present (not shown in the diagram): one mixer, whose inputs are the outputs of the other channels and external inputs: a section with a graphical pre-amplifier equaliser with seven sub-bands; a final integrated amplifier with 50 watt output power. These components, specially designed and realised at IEI, were hosted in the rack of the digital part and inserted below the control panels of the digital component.

Notes on the main functions of the TAUMUS

TAUMUS was designed and realised at CNUCE in large part by P. Grossi. It is a set of procedures to process the musical structures of a piece, through a set of high level commands that can be called from the console of a VM (Virtual Machine) of the IBM computer. As soon as the parameters of the sound to be produced are ready, the program converts them in a format that the TAU2 can interpret. They are sent to the audio terminal with the timing information managed at the physical level by the communication protocol, complying with the durations required by the execution of the notes. Because of the type of sound synthesis used by TAU2, the computation required to the computer is always shorter than the duration of the sounds. This fact always assures real time operation.

The user controls composition and elaboration by various means, which we can describe as follows. TAUMUS has three major functions: *Composition, Re-elaboration, and Library Management*. The programme uses two work areas: the first, called Working Area, is the portion of the memory where all processing on the musical structures is carried out and at the end of which the parameters of the piece are sent to the second area for archiving, called the Library, and, on request, to TAU2 for performance.

The Composition function contains various options: the Text command for transcription and input of traditional texts (see Fig. 5 [p. 111]); the Create command with sub-environments for the guided generation of pieces by means of a certain degree of control in composing, or the automatic production of pieces taken specifically or ad hoc from the archive.

The Re-elaboration functions make it possible to modify the material present in the working area (command Modify), by specifying with various options the zones, voices of the piece, and parameters to treat. The command Vary can carry out a series of modifications in an automatic or casual manner and allows for immediate execution on TAU2 with no time limit.

[See Fig. 5 at p. 113: Levels of codification of the musical text: a) pentagram; b) at the console; c) work area, d) codes sent to TAU2]

The set of commands for Modulation that applies variations to all the parameters (F_ji, A_ni Vi) of a given piece, based on 'models' that can be called up from the archive or defined at the moment, is interesting. To execute the command, the models are placed on special tables in the working area and directly influence the current values, thereby producing modifications in the resulting piece.

In addition to a series of functions typical of archive management, the Library commands make it possible to exploit automatically the performance of pieces with all the other possible functions by using the command Exec. In this way, we can simulate a user who chooses pieces, modifies them and then performs them without interruption. The details of the TAUMUS functions are outlined in Grossi (1976).

Implementation Technology and Construction of the TAU2

We decided to tackle first the design of and experimentation on some new devices for use in the audio unit, which we considered the most critical part of the system. This task was assigned to Chimenti, Ferrucci e Bertini and coordinated by Prof. Denoth. The greatest challenge was to find room for some of the large circuit boards, which resulted from having adopted different solutions to generate signals from those used for the TAU1. In fact, we didn't use the VCO, like most of the analogue synthesisers of the time, to produce the sinusoidal waves and to avoid the frequency instability and, in our case, also because of the high number of signals produced simultaneously. We made instead quartz based square wave oscillators and by using resonant filters LC (with inductors and capacitors) we achieved perfectly sinusoidal and stable the 326 frequencies. The bank of generators was arranged on 36 cards of 20 × 40 cm.

We used the earliest analogue CMOS switches integrated with digital controls available on the market (CD 4009 e CD4010) to select the frequencies specified in the instructions and make them flow to the exits of the selection networks. Since we needed hundreds of chips (each containing 6 of these switches) per channel, it wasn't practical to place them all on one level. We compromised by creating several daughter cards**,** with three chips each, set in larger, reasonably sized mother boards. We also designed new signal amplitude controllers with integrated voltage controlled multipliers, which performed better than those used in TAU1.

[See image at p. 114: Part of a frequency selection matrix with wire-wrap cabling]

Once we determined the dimensions of the logical design of the various parts of the machine, the technical difficulty of the realisation consisted in subdividing the larger circuit parts in smaller sub-blocks, so that they could be contained in boards of uniform size made of printed circuit boards (PCB) with soldered copper tracks**.** Where this wasn't possible, we resorted to wire-wrap technique (more cumbersome and less reliable over time), which involved patiently joining the thin wires with the special tools.

With the construction technology available at the time, IEI's experienced technicians agreed to take on the construction of all of the parts in their laboratories (printed circuit boards, cabling, diagrams, etc.), since it wasn't easy to find this expertise outside.

[See image at p. 115: View of the audio unit (component side): on top 36 boards for the oscillators; in the middle the frequency selection matrix; on the lower part the timbre boards; at the bottom of the rack the stabilised power suppliers]

The design of the digital part and time-sharing system interface, which we considered a more technical problem and less difficult to realise, proved to be much more complicated than anticipated. The choice of the type of connection with the time-sharing system required several meetings with the IBM experts. The electrical and logical specifications required by the IBM interfaces were rather delicate and this meant meticulous analysis of the characteristics in various manuals to evaluate the characteristics of the various peripherals and to choose a suitable connection (use of a fast 4800 modem with serial connection or a more expensive direct cable connection).

In 1973 the fact that the head of the institute recognised in the TAU2 project the potential to best represent the capabilities and competencies in informatics available in the institute at the time, the results of which could then be presented in an important convention to be held in 1975 for the twentieth anniversary of the foundation of the CSCE-IEI helped accelerate work and influenced in part subsequent technical solutions.

Once the project's complexity was recognised[5], the staff was increased and priority was given to TAU2 by all the technical divisions (assembly labs, mechanics labs etc.) Since the most experienced researchers were involved in other projects and able to help with only specific aspects, the institute appointed Chimenti (a CNR fellow at the time) and myself (in recognition of my technical experience and nearness to completion of a degree in the Informatics course) as co-project managers along with Denoth.

With the purpose of speeding up the realisation time of the connection to the computer, it was decided to take advantage of a solution already tested in a previous project between CNUCE and IEI. A direct 16-bit parallel connection between the computer and TAU2 was chosen. A Selector channel of the time-sharing IBM 360/67 in Block-Multiplex mode was chosen. It was managed by a 2701 PDA interface (Parallel Data Adapter Unit-Original Equipment Manufacturer Information, IBM System Ref. Library). The realisation involved high costs, since it needed the deployment of a 150 metre long multi-polar cable between CNUCE and IEI. The cable had to be attached to special hardware interfaces, promising the high reliability, which in fact resulted.

[See image at p. 116: Overview of TAU2: user work station to the left, rack with Digital Unit with equipment for mixing and amplification, and Audio Unit to the right]

Starting up the system was a challenge for everyone involved, especially for me, since I was coordinating practically the entire project together with Denoth and Chimenti, for the analogical part, and Dall'Antonia for the computer connection. During those two years we all worked quickly, with the fear that we wouldn't make the deadline for the convention. In the last few months we worked overtime to finish construction and testing on the digital part and the connection with the computer so that we could execute the first pieces on the system.

[5] No wonder for the long implementation time, because using the then available technologies it took from two to three years and the work of a team of ten people, technicians and researchers, to realise some special purpose computers.

Finally at the Convention ("Venti anni di Informatica a Pisa", "Twenty Years of Informatics in Pisa", June 16-19, 1975), various authorities and international specialists in Informatics were able to hear the first Maestro Grossi's demonstrations on the TAU2-TAUMUS system!

Setting up of hw and sw and experiments with the system

The first tests on the TAU2 essentially involved verifying operational functioning, i.e. that the connection with the IBM system worked properly and that the sounds corresponding to the information sent were correctly generated.

Unlike digital processing systems, the synthesiser produces signal in audio band which have to respect certain quality standards. In fact, the TAU2 audio output produced some buzzing, instability at the lower harmonic amplitude levels (masked up in the beginning by some 'tricks of the trade') and other annoying defects that inhibited working with the necessary precision on the timbre maps and in recording the pieces.

The removal of these problems required another phase of meticulous fine-tuning. The buzzing at 50Hz and 100Hz was largely due to the use of transformers with normal thin plate E + I nuclei for the stabilised power suppliers: since high currents were used (10-15 Ampere) large flux emissions were produced that were dispersed by the iron edges and were captured by the iron structures of the room cabinets and transmitted back to all of the circuits of the audio section. This fact caused disturbances in the audio outputs. A reworking of the entire power supply systems was then necessary. Double C nuclei were adopted, as well as a more accurate stabilisation that took into account the actual peak currents emitted by the system during operation. Further voltage stabilisation was added to the VCA regulator cards to avoid fluctuations in the harmonic amplitude levels. Other interventions involved ground adjustment in many shielded cables of the network for frequency selection.

The Musicology Section of the CNUCE also made improvements and additions to TAUMUS. Modules written in assembler were inserted in the program which was written largely in FORTRAN; the archive of musical pieces and timbre models was enlarged; the procedures for analysis and automatic generation of pieces was improved (Grossi, 1976).

Lastly, after having also regulated the amplitude levels of the formants to avoid saturation when all the selected signals were present, the quality obtained was sufficiently good to begin research on the timbre maps and recordings on disk.

Verification of the sound performance and sound synthesis tests

I've already pointed out that the exact imitation of the sound of traditional musical instruments was not among our priorities for TAU2. Each of us would have programmed the system within the ranges available to obtain sounds with the closest timbres to what s/he wanted. In that sense the terminal + computer was a new, versatile instrument, characterised by a broad and finely tuneable sound space, yet always within certain predefined limits. Nevertheless, we had to carry out certain tests to assess the ability of TAU2 to synthesise and reconstruct signals with certain characteristics. This aspect, which had been anxiously left hanging, was then tackled as soon as the terminal was tested, precisely in order to test properly the phonic characteristics.

It should also be noted that research on the analysis/synthesis of signals in general and of audio signals in particular, with the knowledge and equipment at the time, were difficult with analogical as well as numerical techniques[6]. At any rate, Chimenti and I carried out analysis/synthesis experiments on different signals. As an example, let's look at the tests done on a violin chord 'pizzicato', corresponding to DO_3 (middle C): a sound with a sudden attack, best suited to highlighting the limitations of TAU2.

To carry out the analysis and extract the sound parameters, we proceeded in two ways:

a) analogical way: a sound, with a duration of about one second, was recorded in an anti-reverberation room and then transferred on a circular analogical tape. In this way it was possible to read it continuously on a tape recorder. By using analogical band pass filters, a specially built envelope tracking circuit and an oscilloscope with high persistency tracks, the amplitude envelopes of the partials (integer multiple frequencies of the fundamental) were obtained one after the other, and were reported on graphs and tables (Flanagan, 1965).

b) numeric way: a method based on the DFT was employed, by means of a HP2100 computer in the meantime acquired by IEI and devoted to the multichannel analysis of bio-medical signals. It was equipped with high performance amplifiers and converters: with such an instrument, properly programmed and using appropriate weighting windows, the signal spectra were estimated in short time steps.

[See Fig. 6 at p. 118: Analysis and synthesis of violin pizzicato: a) harmonic spectrum of the first part (duration 5 ms) at the onset b) Progress of output signal from TAU2 over time: phases of attack, decay and beginning of sustain]

Even though the analytical tests were not conducted in an entirely rigorous manner, the results of the two procedures were basically in agreement. We succeeded in tracing the envelopes for the entire length of the sound up to the seventh harmonic with no problem. For the higher ones we were able to estimate the presence up to the fifteenth, with a certain amount of uncertainness, distinguishing an amplitude level from noise only in the attack phase in the first 10-15 ms.

Afterward a sequence of sound and timbre, instructions were introduced for the C3 note (F_{109}) with the amplitude level parameters chosen as to better approximate

[6] The A/D converters were expensive especially in case of meeting the requirements for quality audio and it was difficult to reach a sampling frequency of 32KHz and a resolution of 14-16 bit. These constraints will be overcome with the introduction of new technology only at the end of the 80s. At that time the over-sampling techniques and devices based on the sigma-delta conversion, initially aimed at CD audio readers and later available on chip in trade in the '90s, when the 44.1 KHz e 16 bit resolution standard, still valid today, were introduced.

the spectral data. The fundamental and the first six harmonics were regularly coded in a channel. The other eight harmonics were instead specified in the other two channels with a trick: a voice was associated to each harmonic nf (F_{109}) with n between 8 and 15 that we wanted to add to a sound. The voice frequency had a value the closest possible to the desired one and zero amplitude but the A1 (A1 = A1, A2…A7=0; see Fig. 5 [p. 118] a for an interval of the attack phase). With the metronome base time T at 5 ms, to better simulate the attack phase, the sound produced was satisfying and essentially similar to a string pizzicato.

For simulating a signal well with highly a-periodical components (mainly during the first part of the attack phase), noise + filters generators would have helped. Such an option would have required the addition of other circuits (which were never realised), whose control required special instructions; for instance the R bit in the first byte of the Timbre instruction could have been used as code extension.

On the base of these experiences a set of formant amplitude dynamic deployments was built in co-operation with the researchers of the Musicological Section (Milani, 1976) (an archive of about one hundred timbres); the timbres could have been recalled either according to the phonic needs of a piece or selected in a random way, thus verifying later the sound result.

Assessment of sound performance conducted by outside evaluation

As a result of a comparative study of the various computer music systems in use at that time in Europe conducted for a graduation thesis in Physics (Nencini, 1976), the TAU2-TAUMUS system proved among the most reliable and suitable for the production of Computer Music, including the radio-television corporation (Nencini was an employee of the RAI).

The assessment took into consideration various criteria, among which were: simplicity of language – no need for technical mediation – ability to synthesise complex structures and to elaborate in real time – ability to analyse and deal with external signals – archiving of models and scores – operation time, etc. Compared to the other systems, only aspect missing in TAU2 was the lack of a feature for analysing signals; but then that had not been one of the objectives of the project, since there were already systems on the market and at IEI which could accomplish that task well.

Positive evaluations were expressed concerning the quality of the sound as, for instance, in the review of the double LP *Computer Music – CNUCE/IEI of CNR*, which appeared in the prestigious periodical «Computer Music Journal» (Review by Thomas Blum, vol. 3, 1979, n. 4, p. 58-59).

Once the system was working, we began an intense period of composition of original pieces, re-workings of classical pieces, conducted principally by Grossi but with the full participation of young composers and experimenters in computer music, as some LP recordings can attest. This isn't the place to enter into the merits of genres and choice of types of music used. However, taking into account the opinions of the critics, some expressed perplexity at times concerning some transcriptions-elaborations of traditional music, while a variety of interesting aspects in the original compositions was noted. For example, Grossi's transcription-adaptation of J.S. Bach's *Art of the Fugue* was considered an unusual and well-chosen mix, in which the precision of the computer performance blended well with the particular structural difficulty of the composition.

The first meeting on Music Informatics in Pisa

After about a year of experimentation on methods of working interactively, of composition, of connecting remotely to Italian centres, the TAU2 –TAUMUS system became both the point of reference for various researchers and operators in the field of musical informatics and the incentive for an exchange of ideas on the situation and state of research in Italy. On February 23 and 24, 1976 a conference organised by the Division of Musicology of CNUCE-CNR was held in Pisa, at which was presented the work of the various groups that had formed in the meantime (Pisa, Padua, Venice, Pavia, Naples, Milan, Pesaro, Bologna) and a delegation of the RAI (Gruppo D.O.C.), the latter for input in updating the RAI Phonology Studio in Milan. Demonstrations in real time with TAU2 were organised and a proposal to coordinate the activities between the various initiatives led to the formation of AIMI, Associazione Informatica Musicale Italiana (Association of Italian Musical Informatics). I was particularly interested in the research of Prof. G. B. Debiasi, coordinator of the Padua group, initially active in the area of voice synthesis, and P. Di Giugno, who presented his proposals for synthesis terminals based entirely on digital techniques (microprocessors not yet in use at that time), monitored by 'mid-sized' computers.

Although that meeting produced only oral presentations, the Pisa convention has since been considered the first Meeting of Musical Informatics in Italy and was followed up by a series of other Meetings at regular intervals of a year and a half, and marked by an increasing number of participants and topics of international stature.

Composition, teaching, and demonstration with TAU2-TAUMUS

Among the many aspects of musicological interest, such as analysis, text processing, sound synthesis, simulation of processes of composition, Grossi gave particular importance to the issue of dissemination, given that in order to promote interest in this new topic, it was necessary, especially in that moment, to launch a patient information campaign at all levels in the various sectors of the music world.

Having attained performance in real time and with good sound quality with TAU2-TAUMUS, it was easier to proceed with teaching and demonstrations thus superseding the considerable inconveniences of the very first experiments with the DCMP.

From 1976 on fortnightly courses called "Applicazioni Musicali" (Musical Applications) were regularly included in the teaching calendar at CNUCE. Students from various musical disciplines were admitted, primarily from composition and electronic music (interested in the operational methodologies in use at the centre in Pisa), along with students from scientific disciplines more interested in synthesis methods, circuit solutions and other aspects related to informatics.

The courses included the following topics:

– introduction to the possible applications and trends in computer music
– experiments and research directions at the various research centres at the time
– description of the TAU2-TAUMUS system and instructions for use
– examples of elaboration and creation of sound structures
– short sessions of extemporaneous composition by the students

The most talented and interested in composition were helped to develop original works, presented subsequently at various competitions, national and international events, and included on a few recordings overseen by Maestro Grossi for the musical part and, to some extent, by myself for the technical part.

The courses were held at IEI in the room that housed TAU2, which could hold about ten people maximum. When necessary (for instance during the Gioventù Musicale Italiana concerts) a university classroom a few meters away from IEI was used, by simply extending the TAU2 terminal console + video connections and the audio output.

For internal organisational reasons within CNUCE, the teaching in Pisa was officially suspended in 1978 and, from that point on, we used remote demonstrations more often. In 1981 an experimental year long course in Musical Informatics was instituted at the 'L. Cherubini' Conservatory, which covered all the various issues pertaining to this new discipline.

The form and content of the *remote demonstrations* used for teaching in music schools, academies, theatres, etc. was similar to those indicated above for the courses, though obviously more succinct given the connection costs. Two types of connections were used to carry out these kinds of activities: one with a data line, with a console-video for the session with the time-sharing system that controlled the TAU2 online; the other was an audio line to transmit the audio signal that carried the output of TAU2's three channels.

In most cases, we used only one telephone line for the audio (because of the high cost of renting lines), so the remote band was very narrow, even though we tried to equalise the telephone channel as much as possible in input and output.

On important occasions when we needed high fidelity to transmit signals from TAU2 in output, we used the RAI services from transmitting in FM or with radio connections in UHF. In that case, the signal route and the auxiliary equipment were usually arranged as in Fig. 7 [p. 122]. The objectives of this activity were carried out and obtained successfully, as the reports on the work testify, both in scope and type of centres reached (Nencini, 1985). A summary concludes that RAI transmission was used in five events, data-audio telephone connections in fifteen, and in some thirty other occasions, recordings on magnetic tape.

[See Fig. 7 at p. 122: Components and connections for the use of audio connection via radio with support from the RAI]

Here is a list of some of these events:

- RAI I° TV Channel programme *Anche questa è musica*, with an article on TAU2 by Luigi Fait, in «Radiocorriere TV», n. 1, p. 82-86, January 1976
- demonstration-concert at Accademia S. Cecilia, Rome, data network and audio via RAI, 1977
- two days of permanent Performance, data network and audio via RAI at 41° Maggio Musicale Fiorentino, 1978
- works on tape presented at the Como Festival and at the Premio Italia (Italy Prize) by Gruppo DOC of the RAI, devoted to the use of electronic equipment for producing music in 1977 and 1979
- *Computer valzer*, tapes, Opera Season, Civic Theatre, Florence, pp. 437-441 of libretto, 1980
- Practice sessions on the TAU2-TAUMUS system, data network and audio for the Experimental Course in Musical Informatics at the 'L. Cherubini' Conservatory, Florence (from 1978-1983)
- Connection with data network and audio for the Exhibit "Il tempo dell'uomo nella società della tecnica" Biennale, Venice, 1980, in addition to demos for calendar of events, audio on telephone connection was used from Nov. 8 to Dec; 20, 1980 for a remote composition by Maestro Teresa Rampazzi, later recorded in full band on TAU2.

A complete list and details on the procedures for remote connections can be found in Bertini (1982).

Plans to move TAU2 to Florence

Studies were also completed to determine the feasibility of installing TAU2 directly in the Conservatory, in order to make it possible to synthesise sounds locally in Florence and make remote use of the TAUMUS resources (Bertini, 1981).

An early study, completed as part of a graduation thesis in Informatics (Bertini Giuliano, 1976), had involved the construction of a simple interface using one of the first, cheap microprocessors (the National Semiconductors 4bit SC/MP). In practice, it would have substituted the direct parallel 16-bit connection channel with CNUCE with a high speed (for the time), synchronous line at 4800 bit/sec. between Florence and Pisa. Because of various difficulties, such as the cost of renting the line, the possibility of compromised performance by TAU2 due to problems with the access to the IBM time-sharing system, etc., the interface was never realised.

Along with experts at the Musicology Department at CNUCE, I completed a second study, in 1979-1980, which posited a Z80 microprocessor interface with a 'mid-sized' computer, which the Conservatory would have acquired (a Gould Computer SEL 27/32). With this solution the local resources of the SEL would have guaranteed TAU2 sufficient autonomy to make use of a local archive. However, also in this case, we weren't able to proceed due to delays in acquiring the SEL and other difficulties.

A hypothesis for a telematics service for music

To our way of thinking at the beginning of the 1980s, we could undertake two directions to develop musical applications on the computer:

a) use of very powerful systems, like that designed by P. Di Giugno at IRCAM (Institute de Recherche et Coordination Acoustique/Musique – Institute of Acoustical/Musical Research and Coordination) in Paris, furnished with synthesisers based on Signal Processing numerical techniques and capable of meeting research needs and the demands of complex music. Another example would be the large systems later used by Lucas Film Studios to create sound tracks for the film industry.

b) development of personal computer applications-home computers, which, though limited with sound capabilities, would have allowed many users to work independently and comfortably.

So, it seemed that the prospective of working in remote with TAU2-TAUMUS, along with processing of a central computer service, was destined to end gradually. Instead, encouraged by interest in our way of working, demonstrated also by private industry, we felt the need to experiment further with the possibilities telematics offered, especially by making use of a musical archive organised and available for multi-use and with the possibility of local and remote synthesis. We were helped in this by the contributions of existing realities like the BITNET-EARN-NORTHNET network, which already connected hundreds of centres throughout the world, SIP's ITAPAC data transmission network, the availability on the market of personal computers able to handle audio signals and, in particular, to synthesise musical signals, and the many features of TAUMUS, initially designed to work together with TAU2 and increased to such a degree as to make it adaptable to multi-use and suitably set up remote, music work stations (Nencini, 1986).

A first series of experiments were limited to the Florentine area – between the Conservatory – IROE/CNR-CNUCE, with the equipment available in each of the three centres. Another official experiment of the service was carried out in June 1985 at the event "La luce a Venezia" where a Commodore 64 (with modem), as terminal for remote data from the time-sharing system in Pisa, and as sound 'performer', configured to simulate in some way the TAU2 philosophy and making use of its synthesis hw (Tarabella, 1985; Bertini, 1986), was used as the remote station. Later this system, used for other connection trials, remained a prototype for several reasons (lack of transmission reliability on the switched line and change in work objectives).

Developments in music informatics in Pisa and Florence

About ten years after its construction, various malfunctions began to emerge in TAU2 with increasing frequency, due to obsolescence and defects in the materials, which began to compromise the performance.

The large boards for selecting frequencies, made using wire-wrapping, were the first to show operating defects. For problems caused by mechanical stress (shortage) the wire isolating cover would break at the sharp edges of the pins on which the wire passed. This meant frequent malfunctions and operational defects.

Then another type of defect began to appear on the printed circuit cards, too. The connectors of some of the cards, especially those taken out and inserted repeatedly during trials and/or attempts to find defects, began to make false contacts. Despite restoration of the 'gilding' on the male contacts and the substitution of some of the female contacts (not a simple operation because of the many wires to remove and solder), difficult problems kept occurring.

The system worked more or less reliably without interruption until the end of 1986. In the beginning of 1987, given that both personal and home computers (IBM PC, Apple, Atari etc.) as well as microprocessors for Digital Signal Processing (which we were also working on) were beginning to become more common, it was considered no longer productive to maintain the TAU2 and decided to terminate the activity, which was felt to be more than satisfactory.

Maestro Grossi cut back on his almost daily trips between Pisa and Florence, continuing experiments in musical telematics only (Camilleri, 1988) and then moving his attention to other possibilities offered by the personal computer. Always faithful to his inspirational principles, he developed software procedures for automating composition and graphics processes, using algorithms borrowed from mathematics and physics and launching new paradigms like HOME ART. No so long after he left the work on computer music to his collaborators in Pisa and Florence, according to an agreement between the Conservatory and CNUCE.

Research activity at CNUCE developed then essentially along two lines:

– the first in collaboration with the Conservatory in Florence, where it had started previously, dealt with various aspects of computational musicology and problems relating to the cognitive interpretation of certain musical characteristics (Camilleri, 1991; Leman, 1995).
– the second dealt with algorithmic languages for composition and the development of dedicated systems for executing interactive performances with gestural control, based on using video cameras and infrared devices. The level of creativity in this field meant that a workshop could be organised in Pisa in 1991, with the proceedings published in the journal Interface (Journal of New Music Research) (Tarabella, 1991).

In addition, besides the collaboration on the hardware elements of the systems proposed at CNUCE (Tarabella 1989; Bertini, 1991), research began again at IEI on the use of additive synthesis using VLSI circuits to generate a large number of sinusoids (Barutti, 1993; De Bernardinis, 1998). Systems based on DSP processors were also developed both for the elaboration of musical signals and in general for the audio band (Bertini 1995; 1996), by starting joint partnerships with private firms involved in the production of audio processors.

Conclusions

The operating characteristics the TAU2-TAUMUS system proved to be particularly suited to various aspects of musical informatics, such as composition based on automatic generation and/or guided algorithms, management and performance of pieces from music archives, teaching, musicology, remote demonstrations, and all the applications facilitated by working in real time in connection with the powerful IBM systems. In truth, as with all human endeavours, there has been much light but also some shadows, as for example, limited use of the system on some aspects of psychophysics research and the study of timbres, in part due to the untimely death of one of the researchers at CNUCE, who was the catalyst in those areas.

At any rate, the experience of designing the TAU2 and working with Maestro Grossi contributed positively without a doubt to orienting and promoting research in the work groups at CNR in Pisa and the Conservatory in Florence, on certain peculiar aspects of Musical Informatics and with results recognised world-wide.

In particular IEI, which was rejoined ultimately with CNUCE in one institute (called ISTI-CNR), carried on profitably research on specific topics in musical informatics, following the development of technologies applied to musical signals and addressing, in addition, problems of electro-acoustics. On the other hand, we can see that the techniques for treatment of musical signals were, and are even more now, basic to applications in many other fields, for some of which our institutes were a point of reference, both for public and private research agencies. Furthermore, the many graduation theses proposed and carried out by our group, regarding hd and sw systems based on processors for DSP applied to music, contributed to the training of many graduates involved in teaching, research, and the electronics industry on topics like or related to our own.

APPENDIX

During the 1970s when the TAU2 was under construction, signals in audio band were synthesised for the most part with analogical electronic equipment. The basic circuit was the *oscillator*, used like a modular element to generate various types of wave forms (square, sinusoidal, triangular, etc.), which then added together, filtered and manipulated in various ways, made possible a certain variety of timbres. Later, numerical ways of synthesising were introduced, which had the advantage of greater flexibility and the virtue of adhering better to physical-mathematical and perceptive models. The following is an overview of some of the basic concepts applied to musical signals, useful in understanding the technique of additive synthesis used in the TAU2. For further more detailed information on the various synthesis techniques, we invite the reader to consult the available literature (De Poli, 1983).

We know that a periodic signal can be thought of as the sum of an infinite number of sinusoids with amplitude, frequency and appropriate phases. This fact is basic to techniques of analysis and synthesis of sounds based on the Fourier's transform. In the case of real signals, the sinusoids amplitude tends toward zero as frequency increases. From a practical point of view, we can assume then that a periodic signal can be reduced to a finite number of sinusoids, called harmonics, each of which has a multiple frequency of its fundamental frequency, the latter equal to the inverse of the period of the signal. If possible, such techniques are also used in the case of signals that are not perfectly periodic, by subdividing the signal into successive intervals of time and applying Fourier's transform to each interval. In this way the signal is associated with as many Fourier series as there are intervals into which the signal is divided.

One of the first ways of characterising musical signals is to refer, then, to their degree of periodicity. In this sense, we can define two types of signals: *almost periodic* or harmonic signals and *non-periodic* or non-harmonic signals. The sounds produced by most orchestral string, wind, etc. instruments are of the first kind; while those generated by percussion instruments, for example, are among the second type.

Another way to characterise musical signals is to refer to the evolution over time of amplitude envelope (the line that connects the maximum levels of the sound wave during its deployment). Simply, in each note generated by a musical instrument, we can distinguish at least three consecutive phases: attack, sustain, and decay**.** The duration of each phase and the temporal evolution of the signal in each phase are distinctive characteristics of the various sounds. The attack shows an initial phase with rapid variation, to which correspond many wave components with fluctuating frequencies and rapidly increasing amplitudes, until the sustain phase is reached, when right before the highest harmonics generally fade out. In the sustain phase the wave form is almost periodic with slight variations in its components. In the decay phase the intensity of the signal dies out more or less quickly to nothing. In the case of 'sustained notes', i.e. those played for a relatively long time, the sustain phase is predominant from the point of view of energy. And vice versa in the case of 'staccato' notes and short notes, the attack and release phases are more important.

If we accept a certain degree of approximation, also justified by the characteristics of auditory perception, we can assume that each phase can be subdivided into small temporal intervals of appropriate value, within which the signal can be considered almost periodic. It's possible to apply Fourier's harmonic analysis to each interval and, by resolving in some way the problem of discontinuity at the junctions, we can represent a signal S(t) by a Fourier series sequence, each associated with an interval, as in the following equation [see at p. 128].

If we assume that all intervals have the same duration T and that k is the index that numbers them, the sum of all intervals with index k gives the total time, from the beginning of the signal up to its expiration.

A sound with certain features can be produced by joining a sequence of sections of proper duration. Each section is obtained by summing up sinusoidal signals specified by parameters f, A_n and phase, defined for each interval in such a way to give the sound the desired character. A hardware filtering and special care in the choice of the values to assign via sw to the parameters must be adopted to reduce possible problems at the junction points of the sections.

TAU2's main task was to perform computer music in real time, with a certain degree of polyphony, to vary the parameters of timbre within a vast range of values (plus other functional requirements, as specified above.) It wasn't necessary that the resulting sounds be 'identical' to those of traditional instruments. Considering this fact and acknowledging cost factors, we adopted the principle of harmonic synthesis as expressed in the equation (1), by introducing the following simplifications: M was limited to 7 for each channel (it's possible, however, to reach 15 by using all channels); la φ phase of the signals is omitted, given the conclusion that this parameter is barely perceptible, at least within our synthesis model; all the verifiable parameters (f, A_n , T) are defined by a level number and within a reasonably contained range of values, chosen for simplicity of realisation. In order to generate noticeably non-periodic signals**,** we hypothesised the use of noise generators, for which we would have had to formulate appropriate control instructions. However, this option was never implemented.

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