

Rapports IRCAM



Gerald Bennett

Research at IRCAM in 1978

19/79

Centre Georges Pompidou

IRCAM

Institut de recherche et coordination
acoustique / musique

31 rue Saint-Merri

75004 Paris

Téléphone 277 12 33



Centre Georges Pompidou

Introduction

The year 1978 was IRCAM's first full year in its new building. Both personnel and equipment were brought up to their full complement. The Espace de Projection, completed nearly a year after the rest of the building, was opened to the public in October. The first Session de Formation for composers was held in November and December. Within this framework of rather public events, musical research was pursued in a less visible but most active way.

As in 1977, research at IRCAM in 1978 was mainly concerned with questions touching on the structure of the fundamental element of music, the sound itself. In the previous two years, most sound synthesis and processing was done on IRCAM's large, time-shared computer. In 1978 there was a clear shift of interest toward systems having much of the richness of the multi-user computer but without its two main disadvantages: slowness, and impossibility of use as a concert instrument. In the Electroacoustic Department, led by Luciano Berio, Giuseppe Di Giugno designed and built a portable machine able to synthesize sound in real time, which can thus be used not only as instrument for composition in the studio, but also as a musical instrument in concert. This machine, itself a special purpose computer devoted solely to generating and processing sound, is controlled by a small general-purpose computer which provides instructions for the machine's actual operation. Di Giugno's machine made its public debut in the concerts for the opening of the Espace de Projection in October.

Parallel to the construction of a physical device for sound synthesis in the Electroacoustic Department, a programming language to allow access to this machine was developed in the Computer Department, directed by Jean-Claude Risset. This language will be the basic means of control of a public terminal to be installed subsequently in the Centre Georges Pompidou. Thus an interested public will have the opportunity to experiment with the generation and transformation of sound using one of IRCAM's synthesizers.

Both the special-purpose synthesizers designed and built at IRCAM and the synthesis languages available on the large computer create sounds by combining simple elements to form a final complex result. Such a procedure puts extreme demands on a composer's knowledge of the physical structure of sound and in some sense runs counter to his usual mode of working, which is rather to order and give meaning to a pre-existing material. An alternative form of synthesis is offered by a project of speech synthesis carried out by Xavier Rodet in the Diagonal Department, directed by Gerald Bennett. Here a small computer synthesizes speech directly from typed text. The synthesis is very rapid because most of the information needed to create the actual sound wave is stored in the computer in the form of phonetic elements having a complex acoustic structure. After a cursory analysis of the text entered, the computer calculates the transitions between these elements and then assembles them, rather than creating the wave form from its very basic units. This synthesis scheme is being generalized in order to accept and store non-speech information as well, so that after defining his own sonorous universe the composer can devote a larger part of his attention to composing relations between these elements than to the actual fabrication of sound.

Another important area of research during 1978 has been room acoustics. The completion of the Espace de Projection has provided IRCAM with a unique acoustical laboratory. This space of over 5000 cubic meters' volume has a reverberation time which can be varied between about one half and four and a half seconds, a range never before attained in one and the same space. In addition, the timbre

of the sound in the Espace can be controlled by regulating the proportion between panels absorbing high frequencies and those absorbing low frequencies. An exhaustive series of measurements was carried out in the Espace this year; the analysis of these measurements confirms that important modifications must be made to some of the classical formulae describing the behavior of sound in closed spaces. The redefinition of the mathematical representation of reverberation time and of the formulae for its prediction is a research project for 1979.

In a similarly theoretical way, a series of important studies by James A. Moorer investigated the simulation of acoustical spaces by digital techniques, a question of interest to all composers using digital sound synthesis. Moorer gave a proof and explanation of the inadequacy of the classical techniques of simulating reverberation and proposed an alternative solution.

During 1978 much energy was devoted to the development of existing tools. Perhaps the largest effort has been in the installation of the studios where composers will work and where the tasks of recording and transmitting sound are carried out. In particular, a control studio linked directly with the Espace de Projection was completed, making of the Espace de Projection probably the acoustically most flexible recording studio in the world. Details about the configurations of all the IRCAM studios will be found in Appendix II of this report.

Another large investment during 1978 was the development of the multi-user computer system, employed by virtually all the researchers and musicians at IRCAM. Important improvements were made in the system itself, allowing music synthesis to proceed without impeding other work. In addition, the large sound synthesis program Music V was extended to accept recorded sound and to make transformations upon it. Much of the other development work, too varied in its detail to be mentioned here, took place in the area of computer programming. Details of the computers used at IRCAM will be found in the Appendix I to this report.

Another kind of developmental work was realized in the Instruments and Voice Department, headed by Vinko Globokar, where a new flute is being built which will facilitate the realization of many of the effects called for in contemporary music, without sacrificing the richness and strengths of the classical flute.

IRCAM could only offer its facilities to guest composers in the second half of 1978, after both equipment and personnel were complete. Between June and October three important works were realized at IRCAM: "Wellenspiele" by Balz Truempy, "Arcus" by York Hoeller, both commissioned by IRCAM for the opening of the Espace de Projection, and "Mirages" by Jean-Claude Risset, commissioned by the Festival of Donaueschingen. Each of these compositions used quite different techniques. In particular, Truempy's piece used the machine constructed in the Electroacoustic Department together with live instruments. Both other works employed instruments and a four-track tape synthesized by computer.

During 1978 several projects, notably the speech synthesis project, were financed partially or entirely by the D.G.R.S.T., for which IRCAM is particularly grateful.

IRCAM is also very grateful for both the moral and financial support given by the IRCAM Foundation, and particularly by its President, Paul Sacher.

During the year IRCAM received guest researchers for visits of varying duration from the following universities: Brussels, Goettingen, Hamburg, London, New York at Buffalo, Paris V, Paris VI,

and Stanford; from the Massachusetts Institute of Technology, the Royal Academy of Technology, Stockholm, Bell Laboratories, and the CNRS.

The first Session de Formation for composers took place from 6 November to 20 December, organized by the Pedagogic Department, which is headed by Michel Decoust. During the course of the session, the participants visited the following studios in France: Groupe de Recherches Musicales, Groupe de Musique Experimentale de Bourges, Ecole Nationale Supérieure de l'Electronique et de la Radioélectricité de Grenoble, Groupe de Musique Expérimentale de Marseille, and the laboratories of the C.N.R.S. at Marseille.

IRCAM accepted stagiaires from the Conservatoire National des Arts et Métiers, the Ecole Nationale Supérieure des Télécommunications, the Groupe de Recherches Musicales, and the Ecole Nationale Supérieure de Physique in Marseille. In addition a group led by the composer Pierre Barbaud and one led by the computer scientist Patrick Greussay were invited to do work related to their own projects within IRCAM on a regular basis.

IRCAM published 18 scientific reports during 1978, and IRCAM members were invited to give presentations at professional congresses in Aarhus (Denmark), Chicago, Hamburg, Honolulu, Madrid, and Stockholm.

RESEARCH PROJECTS

Sound Synthesis

The most important electronics construction project during 1978 was the design and construction of a very rapid digital synthesizer by Giuseppe di Giugno and Alain Chauveau in the Electroacoustic Department. Already in February, 1978 di Giugno had realized a definitive version of a device designed and built in 1976, and soon christened the "4A Machine". The 4A Machine is a digital synthesizer of 256 oscillators whose frequency, amplitude, and phase are under program control by the user. The oscillators cannot be interconnected. In April, 1978 di Giugno and Chauveau completed a much more supple and sophisticated digital device for the synthesis and processing of sound named the 4C Machine (the Annual Report for 1977 described in detail its predecessor, the 4B Machine).

Although the design of the 4C Machine is optimized for sound synthesis, in fact the device consists of four very fast micro-modules, an adder, a multiplier, a "modifier", whose input is modified by a stored function, and a logic unit. These modules are multiplexed to give the following basic structural units: 64 oscillators, 92 multipliers, 32 logic units, 64 modifying units, and 32 timers. The operations and interconnections of the micromodules are fixed by microprogram, and it is the particular structure of the microprogram stored in read-only memory which is responsible for the remarkable speed of the 4C Machine. These connections are thus inaccessible to the user, although they can be changed if so desired. The interconnections among the basic units however are completely under the control of the user and are defined by program.

Although theoretically the number of possible interconnections between the basic elements is vast, certain combinations of elements recur again and again for the synthesis and processing of sound. To make these combinations more easily available, di Giugno, together with Jean Kott, began the development of a series of subroutines to represent the configurations of the machine most frequently used. These subroutines can of course be combined with each other or with the basic units to form more complex configurations capable of being called by a single command. From the earliest experimental phase on, di Giugno has felt it important to avoid conceiving general programs of control for the 4C Machine, preferring to let the experience gained through the use of small, specific functions and the musical needs which become evident through contact with the machine itself determine the shape a later control program will take. The final system will consist of many different elements of complexity similar to that of the 4C Machine, but having different functions (reverberator, digital mixer, etc.), and the general control program will need to reflect the requirements of all of these individual elements as well as those of the control operations for the entire system.

The first public performance of the 4C machine took place in the concerts for the opening of IRCAM's Espace de Projection in October, and again a week later at the Donaueschingen Festival in Germany, where it was used in the work "Wellenspiele" commissioned by IRCAM from Balz Truempy, a young Swiss composer. Truempy's work required that live instruments control and direct the synthesis by the 4C Machine. Since this first prototype did not allow the input of actual sound, envelope followers were used to provide signals to the machine. In a later version of the 4C Machine, digitized real sound can be monitored directly for control information.

For the concerts in the Espace de Projection, the 4C Machine was attached to a PDP-11/55 computer in the Electroacoustic Department. A display terminal, the envelope followers, and several manual controls were in the Espace and relayed their signals to the 4C Machine via permanent connections between the Espace and the rest of the building. For the concert in Germany, Digital Equipment Corporation (Germany) very kindly lent and installed a PDP-11/40 computer in the Donaueschingen Sporthalle, where the concert took place. The 4C Machine itself is eminently portable.

The programming for the piece by Truempy was done by Giuseppe di Giugno and Jean Kott. Instrument design was done by Neil B. Rolnick together with Balz Truempy. Rolnick also participated in the performance.

Whether performed on general-purpose digital computers or on special-purpose devices, most sound synthesis at IRCAM is carried out by linking together simple modules (oscillators, ramp generators, multipliers, etc) to form complex virtual instruments. These modules may be subroutines or hard-wired elements, but in either case they require a subtle knowledge of signal processing and the ability to imagine and manipulate sound structures from their smallest unit up, both of which are rather rare among composers. An important alternative to this conception of synthesis has been developed at IRCAM during the past year by Xavier Rodet.

This sound-synthesis system is derived from a system of speech synthesis-by-rules developed by Rodet at the Service d'Electronique of the Centre d'Etudes Nucléaires at Saclay. This system accepts normal (French) text input at a computer terminal and calculates by rules certain acoustic parameters (fundamental pitch, formant pitches and amplitudes, as well as the duration of the present pitch period) for each pitch period of the given phonetic string, taking into account certain prosodic features. The acoustic signal is formed by the summation of formant time-domain functions derived from the previously determined parameters. The calculation of the signal takes no longer than the time required to speak the particular text example.

The phonetic transcription of a given text is carried out by a set of rules embodied in a program. These rules attribute the customary pronunciation to each alphabetic letter within a given context. The set of rules is supplemented by a small exception dictionary (approximately 500 words) and is being tested on a French dictionary with 65000 words. The sentence is divided into "prosodic clusters" by use of a list of what Rodet calls "grammatical function words" (articles, pronouns, pronominal adjectives, propositions and conjunctions). The general fundamental frequency contour is calculated as a decreasing line from the beginning to the end of the sentence. Along this line is superimposed a rising pattern for each non-terminal prosodic cluster. The terminal group is given a decreasing frequency contour.

The calculation of the acoustic parameters is done for each pitch pseudo-period in voiced sounds and each segment of approximately the same duration in unvoiced sounds. First of all the program determines a certain number of features of the phonemes (such as nasality, occlusion, friction and positions of articulators) in the surrounding VCV or CVC context. Then the fundamental frequency and the frequencies and amplitudes of a variable number of formants or noise components are calculated by rules with respect to these features and with respect to the phonemes of the VCV and CVC context.

The signal is computed period by period (or segment by segment in unvoiced sounds) using a summation of M ($M < 7$) time-domain functions, each corresponding to one or more of the formants. These functions are derived from recorded speech in the case of unvoiced sounds. For

voiced sounds the system uses four tabulated functions derived from a linear predictive coding model of one pitch period of natural speech. The calculation of a wave form is initialized at the beginning of each pitch period and is continued until it is sufficiently attenuated during the following pitch period, where it is added sample by sample to the new wave form initiated at the beginning of the following pitch period.

Rodet's research at IRCAM during 1978 has been directed toward two questions: the improvement of both the algorithms and techniques for speech signal construction on the one hand, on the other the modification of the system's design to allow it to treat more general acoustical information.

Control Languages for Sound Synthesis

During 1978 Philippe Prévot wrote the first control language for one element of the sound generating and processing system being developed at IRCAM by Giuseppe di Giugno. This device, known familiarly as the 4B-machine, is described in detail in "A One Card 64 Channel Digital Synthesizer" - G. di Giugno and H. Alles (IRCAM paper 4/78). It is connected to a Digital Equipment Corporation PDP11/03 computer using floppy disks and is designed to be part of a portable system for use in both studio and concert hall. In the present configuration the computer is also attached to a real-time input device consisting of 48 sliding potentiometers, a 61-key keyboard, a pedal, a two-dimensional wand, 16 movement-sensitive potentiometers and 32 buttons.

The design of the control language is based very closely on that of the 4B-machine itself. The characteristics of the machine can be summarized thus: it is a digital sound synthesizer with 64 multiplexed oscillators capable of frequency modulation and operating at 32 kHz sampling rate, 128 ramp generators operating at 4 kHz, and 15 general registers used to interconnect oscillators and ramp generators. Whereas an analogue oscillator is a device which is set to a value which remains constant until the next change, an oscillator of the 4B-machine is controlled by two ramp functions, which of course may be constant, but which in the general case are meant to change. To control these ramps requires three pieces of information each - initial value, final value, and increment - for a frequency and an amplitude ramp. Upon receiving this information the oscillator begins playing and has no further need of any control device. When a ramp reaches its final value, it requests more information by sending a program interrupt to the controlling computer. The primary function of the controlling computer is to translate programs and scores into these three quantities - initial value, final value, and increment - for the 4B-machine. In Prévot's language these quantities are called a "plex" and form the primary and principal kind of data sent from the controlling computer to the 4B-machine.

An important aspect of Prévot's language is that it is largely interrupt-driven. Both the 4B-machine and the manual controls send interrupts to the controlling computer which are attended to immediately, stopping whatever else is going on at the moment. When all interrupts have been attended to, the real-time processor drops into the "background", to attend to matters less urgent. The choice of whether a real-time operation will be attended to by interrupt or by the background monitor is largely the user's decision. To the controlling computer all interrupts look the same, whether coming from the 4B-machine or the real-time inputs.

Using the language processor, one must write a program for the 4B-machine and the real-time inputs, consisting of instructions for

the internal connections to be made in the machine and of the definition of any functions to be used. The language can be used in two different modes: either one writes a number of files which can be recalled and played, as would be the case for the performance of a piece, or one can work interactively with the machine, hearing the effect of each new command given at the terminal, adding and deleting text freely, etc. The boundary between these two modes is very flexible, for a simple switch allows the user to make a file of what had been used interactively, or to make corrections to an existing file, deciding only later whether to keep or discard these corrections.

The notation of the internal connections of the 4B-machine requires intimate knowledge of the machine's structure; hence only its flavor can be suggested here. To define the connections for the smallest physical unit of the machine (an oscillator) four things are needed: the name of a register for frequency modulation, if there is to be any, the name of where to find the frequency information (whether a constant, a function or a real-time input), the name of the wave-table generating the desired wave form, and the name of where to find the amplitude information for this oscillator or instrument. This information is routed to one of the 15 general registers of the 4B-machine.

The second part of the program must be a definition of any functions required. Wave-table functions can be read in from disk files and are characteristically 4096 words of 14 bits. Other functions can be defined at the teletype and are either step or breakpoint functions. In particular, breakpoint or interpolating functions must be used to define envelope functions, where one wants to make rise and decay times constant while scaling the rest of the function according to the duration of the event for which the envelope is used.

Besides these two main parts of the program, there are a number of commands which can be used especially for temporal control. One of these is named "metronome" and determines the value of the basic durational unit referenced when a score is used. The metronome command can also be associated with a real-time control or with a function and thus can be varied either manually or under program control during execution. Another command is the "pulse" command, which provides a regular interruption whose duration is likewise determined either by a real-time input or by a function. The pulse function must be qualified further by associating with it an action and something for it to act upon. Several other commands, too numerous to be listed in detail here, are defined as well.

Such a program, consisting of a description of connections, definition of functions, and further commands, is quite sufficient for making music. In this case, control and articulation (pitches, rhythms, durations, timbres, etc.) would presumably come from a combination of real-time inputs and functions. The language also accepts a score, however, allowing the user to specify there any information which he wants to coordinate temporally as a note (or an event, if the instrument is complex).

At the time of writing the language was being used to develop a control terminal for use by the public in the Library of the Centre Georges Pompidou. Philippe Prévot is also carrying out the modifications necessary to adapt the language to the newest element of di Giugno's system, the 4C-machine.

The performance of a version of "Explosante-Fixe" by Pierre Boulez for flute and digital synthesizer planned for May 1979 has hastened the development of a first control language for the 4C-Machine. This language was designed by Curtis Abbott; it was written by Abbott and James Lawson.

The language has two separate parts. The first is a compiler which translates input expressed according to a specific syntax into data, the second, the run-time system translates this data into control commands for the 4C-machine. The compiler is written in Pascal, the run-time system in assembly language for a Digital Equipment Corporation PDP11.

This language differs in important ways from the Prévot language described in detail above. First, it was specifically designed for the PDP11 with large-capacity disk memory and floating-point hardware, whereas the control language for the 4B-machine was designed for a micro-computer system. In addition the language was designed to allow digital recording and playback to be treated notationally and conceptually similarly to other real-time activities.

The structure of the run-time system is very dependent on the physical structure of the 4C-machine, because of the housekeeping necessary to implement instrument definitions, so that once again it is impossible to describe in detail a language without describing the machine it should control. Nonetheless, a brief survey may give some sense of the language.

The compiler and the run-time system of this language can be separated, and in fact the run-time system can serve many different compilers. In the run-time system parallel structures are represented by "chains". The chain is the basic unit of the system. It can be visualized as a two-dimensional matrix whose first column is a list of information about when to start executing the corresponding line, whose other columns are lists of elements to be treated in the same way (e.g. a column of frequency information, another of amplitude information, etc), and whose lines represent events containing elements to be executed simultaneously.

The temporal offsets can be either values (duration of this event) or the instruction to wait for a real-time interrupt to start processing the event. The elements of the chain can be values, instructions to look up a real-time value, instructions to read a value from a file, instructions to start another chain, or one of a series of commands (load or unload an instrument, record, playback, etc.). The possibility of loading and unloading instruments (as well as wave-tables of any length) permits great flexibility in the use of the resources of the 4C-machine. Parallel activities are simulated by having an element of a chain be a starting instruction for another chain. An element can start an instance of any chain, including its own.

The development of a system for speech synthesis-by-rules required a control language, no model for which existed at IRCAM. During the past year, Jean-Luc Delatre developed a language whose importance goes far beyond the control of a specific synthesis system. In the synthesis system developed by Xavier Rodet (see above) elementary components (generally the description of the evolution of formants, but also of broad-spectrum noise) are calculated simultaneously and combined to form a complex acoustical output. A library of subroutines within the language can serve to calculate the elements; complex rules derived from analyses of natural speech determine which of the subroutines will be involved to calculate a particular sample of a particular element. Management of these rules is the task of the language. Since the control language does not know at the beginning of a calculation whether a certain element is constant or evolving (an amplitude may remain the same for some time, or it may change rapidly) or even whether an element will always be present or not (voiced sounds differ in the number of relevant formants they have), it must be able to deal with circumstances which cannot be foreseen at the beginning of a complex

succession of operations.

The acoustical signal is calculated sample by sample by a computation loop containing all the subroutines relevant to the present sample. Elaborate control procedures are necessary to run the loop efficiently, for it must be possible at any moment of computation to include new subroutines in the loop, to remove subroutines temporarily and to put them in a waiting queue, to associate them with the calculation of new elements (formants), etc. This event management is particularly effective for decision-making subroutines whose truth value changes only rarely. These control procedures make of what was originally a control language for a specific synthesis system a language for general purpose simulation of continuous systems. The language developed by Jean-Luc Delatre is described extensively in "Le système de traitement de signaux digitaux 'Junior' (IRCAM paper).

This generality, originally necessitated by constraints of efficiency, makes the language of great interest to musicians, for this reason: the structural relationships at the most minute level (the sample) are represented in precisely the same terms as those governing the evolution of middle- (the phoneme) or large-scale structures (the phrase). Hence the language can actually function as a compositional tool, allowing the formulation of precise relationships while leaving some freedom as to the material to which to apply these relationships. In this sense, the language differs fundamentally from languages like Music V, which are essentially the translated description of musical processes worked out in a different medium. Although there is no musical "syntax" proposed by the language, it certainly leads itself to the formulation of individual syntaxes for composition.

Acoustics

The most important work in acoustics during 1978 was James A. Moorer's study of reverberation, "About This Reverberation Business" (IRCAM paper 17/78). The main concern in Moorer's work was to find an improved algorithm for simulation of room reverberation by computer, a subject of obvious interest to musicians.

The research was done in three parts. The first part was a detailed investigation of the classic technique to simulate reverberation on the computer: the use of recirculating delays with different gains. These delays arise from a series of digital filters, connected either in series (all-pass filters) or in parallel (comb filters). Room reverberation has been simulated on computers at least since 1961, but the results have usually been disappointing for musical use: among other faults the decay of most digital reverberators has a metallic sound, and reverberation of short percussive sounds tends to sound more like flutter echo than real reverberation. Moorer developed a number of new unit reverberators, and he proposed and tested a reverberator consisting of six comb filters in parallel, each with a low-pass filter (to simulate the attenuation of sound caused by air) in the loop, which gave results quite comparable with the reverberation of real rooms.

In the second part of his study, Moorer investigated simulating reverberation by using a simplified impulse response of a real or imaginary room. While it is always possible to simulate the reverberation of an actual room by convoluting the room's impulse response with an acoustic signal, the procedure uses disproportionate amounts of computation time (six minutes for one second of sound in Moorer's study, using very fast algorithms). Moorer attempted to arrive at more reasonable computation times by using a simplified impulse response, containing many fewer points than the full

response. The examples synthesized using this method did not sound very realistic. Moorer hypothesized that this is because the model does not take into account the effects of diffusion, which add several orders of magnitude to the impulse's complexity. Further work with this model showed why recirculating delays will always be inadequate to simulate real reverberation. The details are too complex to discuss here, but the demonstration of this inadequacy is important for all who use artificial reverberation.

In the third part of his study, Moorer proposed a reverberator using an actual impulse response convoluted with the unreverberated sound for the first 40 to 80 ms. of reverberation. Recirculating delays would be used for the remaining time. This model gives good results and represents a savings in computation time of a factor of 10 to 20 compared with straight convolution.

In another study, "The Digital Coding of High-Quality Musical Sound" (presented at 62nd Convention of the Audio Engineering Society, Brussels, March 1979, IRCAM paper), Moorer investigated two different techniques of data reduction for the digital encoding of sound: minimum length encoding, or Huffman coding, and an incremental floating point encoding scheme which Moorer developed.

The first scheme offers in general a substantial reduction of the number of bits necessary to represent sound (five to seven for a 16-bit dynamic range), but it also has two important disadvantages: under certain circumstances it can actually augment the data up to a factor of two, and its irregular storage patterns may present problems. Moorer's floating point coding is capable of reducing a 22-bit signal to 13 bits (mantissa of nine bits, exponent of four) with very little perceptible loss of quality. This means a three-bit reduction over the 16-bit integer sample generally employed for high-quality sound processing.

DEVELOPMENT OF EXISTING TOOLS.

During 1978 a number of important improvements to the PDP10 systems were made. The demands made on a computer used for sound processing and synthesis, where it is important for the tasks dealing with sound to avoid being interrupted at certain moments, are often not compatible with those made on a system time-shared by many users. For example, at IRCAM the system program PLAY used to output digital sound files demanded exclusive use of the computer for the duration of each sound file played. The program used very little computer time but it protected itself against interruption by other users when it was not actually inputting data from the disk by being of higher priority than any other job and by looping continually until the entire sound file had been transferred from disk to converter. Jean-Louis Richer and Raymond Bara changed PLAY so that it no longer loops but rather defers to other jobs while waiting to read in data from the disk. The time constraint on the program was very severe, for at the high sampling rates necessary for good quality sound there was very little time for a controlling program to do the continual housekeeping necessary when many tasks are being moved in and out of central computer memory.

Bara has also developed a group of programs allowing the use of the electrostatic printer at IRCAM both for text and for computer graphics. Besides the two programs for printing text and graphics, there is a program allowing each user to design his own fonts, and there are programs for preparing documents, indexing, etc. At the moment the electrostatic printer cannot be used from every terminal, as can the line printer. The development of a software infrastructure to make this possible is a project for the future. This report was entirely prepared and reproduced using the editing and printing aids of the system.

Among the more technical tasks of system development, it is important to mention the conception of a large "swapping memory" of 512,000 words to shorten user waiting time when many tasks are moving back and forth from disk to central memory, and the design of a 24-channel direct memory access, which will greatly alleviate the load on the system when recording or playing sound. Both devices will be built in 1979.

The development of Music V continued during 1978, through the efforts of Jean-Louis Richer. The most important addition was a series of unit generators allowing the treatment of real sound. Sound files already recorded on the computer can be used like any other input to a unit generator. In addition sounds can be read at varying speed, from any place in the file, backwards or forwards, with the precision of one sample. Other unit generators permit delaying one voice in constantly varying relation to another, again with the precision of one sample. A complete documentation of Music V as used at IRCAM has been written by Richer.

An important project during 6 months of 1978 has been the construction of a new flute, a project largely supervised by Robert Dick. The main goal of the project is to render the flute more easily capable of answering the demands of contemporary music, particularly as regards multiphonic sounds. On presently existing flutes multiphonics can only be transposed to a limited degree: because of the mechanical connection of many keys in the Boehm fingering system one can only reproduce the required pattern of open and closed holes until reaching a key which closes more than one hole at a time. The flute under construction will disconnect mechanically connected keys, while spacing the holes around the body so as to reduce to a minimum the strain of the hands to reach them. All 10 fingers will be free to play. Certain "compromise fingerings" in the Boehm system (particularly e6 - f#6 - g#6) will be corrected.

A first prototype of a flute designed for playing multiphonics was proposed and calculated by Arthur Benade, member of the Scientific Committee of IRCAM, in June 1978. In this prototype the player had individual control of each hole, and all the holes were small (on the Boehm flute there are both large and small [half-closed] holes; the large holes ensure a full sound, but make it more difficult to let the air column vibrate in more than one mode at a time; it is this multi-mode behavior which causes multiphonics). In the present design it was decided to keep the principle of independent control of each hole, but to choose a larger boring to avoid sacrificing the fullness of the sound.

The flute has been realized with the collaboration of Alfred Cooper, London, who calculated the positions of the holes and resolved many of the mechanical problems.

In the Electroacoustic Department Maurice Rozenberg constructed a digitally programmable bandpass analog filter. A microcomputer controls the filter's parameters. The functions of variation as well as the microcomputer's program are controlled by a DEC PDP11 computer. The filter achieves attenuation of 50 dB per octave. (See "A Digitally Programmable Filter", IRCAM Paper 16/78).

Appendix I by Raymond Bara

Computing equipment at IRCAM

The following is a brief description of the present computer possibilities at IRCAM, presented from the point of view of a computer scientist. The list of equipment is essentially complete; software is presented only in a general way. The interested reader can refer to [2] for greater detail. In all of the following, the abbreviation DEC refers to the computer equipment company Digital Equipment Corporation.

A. DEC 10 System.

IRCAM's PDP-10 system is composed of:

- a DEC KI10 36-bit central processing unit;
- a core memory with 256 K 36-bit words, composed of:
 - 64 K words of DEC memory (MF 10G),
 - 192 K words of AMPEX extension memory (ARM 10L);
- a disk memory containing a total of 60 M 36-bit words (equivalent to 270 M 8-bit bytes), composed of:
 - 1 DEC RP10C disk controller,
 - 1 DEC DF10 direct access memory unit,
 - 3 DEC RP03 disk units with 10 M 36-bit words (equivalent to 45 M 8-bit bytes) each,
 - 3 AMPEX DM 323 disk units with 10 M 36-bit words (equivalent to 45 M 8-bit bytes) each;
- a magnetic tape system composed of:
 - 1 DEC TM10 controller (connected to the I-O bus),
 - 2 DEC TU10 magnetic tape drives (9-track, 800 bpi, NRZI, 36 K characters per second);
 - one DEC LSP10 line printer (300 lines per minute);
 - an asynchronous line controller composed of:
 - 1 DEC DOC10A,
 - 4 DEC DC10B, which permits the connection to the system of a maximum of 32 asynchronous terminals functioning at a maximum speed of 9600 bauds;
 - 2 channels of 16-bit digital to analog conversion built at IRCAM by David Cockerell;
 - 1 high-quality channel of 16-bit analog to digital conversion built in London by Tim Orr on request from IRCAM (signal/noise ratio > 85 db, dynamic > 90 db);
 - 1 high-speed interface between the PDP10 and the PDP11/34 built by David Cockerell at IRCAM (10000 36-bit words per second).

The terminals presently connected to the system are:

- 21 DATAMEDIA ELITE 2500 video consoles in a non-standard configuration (SUMEX, ROM IMSS keyboard with additional modifications made at IRCAM). Their principal quality is the ability to insert or delete characters in a line without redrawing the whole screen;
- 1 DEC VT50 video console,
- 2 DEC LA36 "hard-copy" terminals.

These terminals are positioned throughout the IRCAM building. The operating console is also a DEC LA36.

B. DEC 11 Systems

1. Graphic system GT44

Composed of a DEC PDP 11/40 mini-computer with a memory of 32 K 16-bit words, 2 DEC RK05 disk units with 1.2 M 16-bit words each, and a graphic display unit with light pen. In addition, a VERSATEC 1200 electrostatic printer with resolution of 200 points per inch is attached.

2. PDP11/34 System

Composed of a DEC PDP 11/34 mini-computer with a memory of 64 K 16-bit words, 2 DEC RK06 disk units with 6.6 M 16-bit words each, a graphic display unit with light pen, one channel of 16-bit digital to analog conversion made at IRCAM by Didier Roncin, an analog to digital converter (Tim Orr), an interconnection matrix (Peter Eastty), a 4C system (designed and built at IRCAM by Giuseppe di Giugno, modified and extended by Peter Eastty and Didier Roncin) with a control unit composed of an organ keyboard and 48 potentiometers scanned by a MOTOROLA 6800 microprocessor which converts their parameters to digital form for use by the computer.

3. PDP11/55 System

Composed of a DEC PDP 11/55 mini-computer with a memory of 64 K 16-bit words, 2 DEC RK06 disk units with 6.6 M 16-bit words each, a graphic display unit with light pen, and a 4C system (Giuseppe di Giugno). The PDP11/55 is used in the development of di Giugno's sound processing system.

4. LSI11/03 System

Composed of a DEC PDP 11/03 mini-computer with a memory of 32 K 16-bit words, 2 DEC RX01 floppy disk units with 118.6 K 16-bit words each, one DEC VT50 video display console, and a 4B system (Giuseppe di Giugno). (cf. [3]).

5. Connection to the PDP10 System

With the exception of the PDP11/55, the mini-systems are connected to the PDP10 by the DC10, and can be used as terminals for the PDP10 as well as independently. With the exception of the 11/03, they can also serve as graphic terminals for the PDP10.

Monitor systems and languages

The monitor used for the PDP11's is RT11, the standard for DEC.

The monitor for the PDP10 is an intermediate version between DEC TOPS10 and the monitor of the Artificial Intelligence Laboratory of Stanford University (cf. [1]), one of the important characteristics of which is the optimization in the use of the Datamedia terminals, permitting the use of the Stanford text editor, E. Many of the utility programs come from Stanford, others are from DEC (compatibility with DEC is maintained). The Computer Department of IRCAM maintains this system and introduces into it modifications which are necessary for musical applications and for its use by people with little training in computer science.

Among the languages available on the PDP10, in addition to assemblers, we can mention:

- ALGOL (DEC)
- BASIC (DEC)
- BLISS (DEC)
- FORTRAN (DEC)
- PASCAL (University of Hamburg)
- SAIL (Stanford's much-enriched version of ALGOL)
- SNOBOL (Bell Laboratories)
- VLISP (Computer Science Department, University of Paris, Vincennes)

REFERENCES

- [1] Martin Frost and Brian Harvey,
The Stanford/IRCAM Monitor..
IRCAM Report, April, 1977
- [2] John Gardner et al.,
Computer Facilities for Music at IRCAM, as of October, 1977.
IRCAM Paper 3/78.
- [3] Giuseppe di Giugno and Hal Alles,
A One Card 64 Channel Digital Synthesizer.
IRCAM Paper 4/78.

Appendix II
by Ben Bernfeld

Audio Possibilities at IRCAM

1. "Small" studios

There are four "small" studios, two in the Computer Department, one in the Electro-Acoustic Department, and one in the Pedagogy Department. They are principally intended for listening to magnetic tapes, listening and recording signals from digital equipment, and sending analog signals to digital equipment as well as to tape decks for copying. The basic equipment for these studios consists of:

- a) 1 Neve mixing console with 12 microphone/line inputs, 4 principal outputs, and 2 auxiliary outputs (for example, usable for artificial reverberation);
- b) 1 Ampex ATR-102 tape deck, 2 track, 1/4 inch, 15 and 7-1/2 inches per second;
- c) 1 Ampex ATR-104 tape deck, 4 track, 1/2 inch, 15 and 7-1/2 inches per second;
- d) 1 DBX 4-channel noise reduction system, each channel of which can be used either in coding (recording) or decoding (reproduction);
- e) 4 loud-speakers and power amplifiers for stereo and quadraphonic monitoring.

The distribution system for these studios allows for ease in interconnection between audio units, in interfacing with digital equipment, and in interconnecting with other studios in the building.

One of the Computer Department studios additionally contains the following auxiliary equipment:

- a) 2 third-octave graphic filters.

The Pedagogy Department additionally contains the following auxiliary equipment:

- a) 1 Ampex ATR-102 tape deck, 2 track, 1/4 inch, 15 and 7-1/2 inches per second;
- b) 1 DBX 4-channel noise reduction system, each channel of which can be used either in coding (recording) or decoding (reproduction);
- c) 2 third-octave graphic filters;
- d) equipment for analog sound processing.

2. Electro-acoustic studio

The Electro-acoustic studio fulfills the same functions and includes the same equipment as the small studios, with the difference that the console, with 28 inputs, is much more powerful. In the near future this studio will be equipped with a complete remote control system permitting use of the 16-track tape deck located in the recording studio of the Instruments and Voice Department.

3. Control room of the Espace de Projection

The control room fulfils the same functions and is composed of the same basic equipment as the small studios with the difference that: a) The Neve mixing console has been extended to permit direct line outputs on each of the 12 inputs. b) The shape of the room permits the use of only 2 loud-speakers, although the quadraphonic capabilities of the equipment are conserved. c) The control room includes a bi-amplification system for the 8 loud-speakers (JBL 4350's) which can be mounted in the Espace. d) The control room includes a complex distribution system which permits interconnection of the 44 microphones which can simultaneously be used in the Espace, as well as connections to echo chambers, sound output lines, a small portable mixer, etc.

The principal function of the control room is to permit the realization of concerts with electroacoustic support or sound reinforcement, although this does not exclude its use as a recording facility for productions not requiring more than 12 microphones.

4. Master recording studio

The master recording studio brings together the finest audio equipment in the building. It is intended for the production of professional recordings on all levels, as well as for research projects which demand an audio facility of the highest quality.

The equipment in this studio consists of:

- a) 1 Neve mixing console with 32 microphone/line inputs, 16+4 principal outputs, and 8 auxiliary outputs;
- b) 1 Ampex ATR-102 tape deck, two-track, 15 and 7-1/2 inches per second, 1/4 inch;
- c) 1 Ampex ATR-104 tape deck, four-track, 15 and 7-1/2 inches per second, 1/2 inch;
- d) 1 Studer A - 80 tape deck, two-track, 15 and 7-1/2 inches per second, 1/4 inch;
- e) 1 Studer A - 80 tape deck, four-track, 15 and 7-1/2 inches per second, 1/2 inch;
- f) 1 Studer A - 80 tape deck, sixteen-track, 15 and 30 inches per second, 2 inch;
- g) 1 Dolby 16-channel noise reduction system, each channel of which can be used to code (record) or to decode (reproduce);
- h) 16 DBX modules which can be used with the Dolby system;
- i) 8 Dolby mono noise reduction systems with Dolby, DBX, or Telcom modules;
- j) 4 JBL bi-amplified 4350 monitor systems;
- k) complex system of interconnection with the facilities of the Espace de Projection, the other audio studios, and the anechoic chamber (21 audio circuits - in preparation).

5. Studio facilities for the Instruments and Voice Department

The Instruments and Voice facility is equipped for professional recordings of small ensembles and for research activities.

The equipment in this studio consists of:

- a) 1 Neve mixing console with 28 microphone/line inputs, 16+4 principal outputs, and 8 auxiliary outputs;
- b) 1 Ampex ATR-102 tape deck, two-track, 15 and 7.5 inches per second, 1/4 inch;
- c) 1 Ampex ATR-102 tape deck, four-track, 15 and 7.5 inches per second, 1/2 inch;
- d) 1 Studer A - 80 tape deck, sixteen-track, 15 and 30 inches per second, 2 inch;
- e) 1 DBX 16-channel noise reduction system, each channel of which can be used to code (record) and to decode (reproduce) simultaneously;
- f) 2 DBX 4-channel noise reduction systems, each channel of which can be used to code (record) or to decode (reproduce);
- g) 4 JBL 4343 studio monitor systems.

In the near future there will be the possibility in the Instrument and Voice Studio of connecting 16 speaker systems to the console in order to conduct acoustic and psychoacoustic studies.

6. Reverberation chambers

A portable AKG spring reverberator is available, but can only be used for tests or sound reinforcement. There are two reverberation chambers in the old IRCAM building, equipped with Quad electrostatic loud-speakers and AKG electrostatic omnidirectional microphones. Connection with the two chambers can be made from Studio 1 of the Computer Department, the studio of the Electroacoustic Department, the control room of the Instruments and Voice studio, the master recording studio, and the control room for the Espace de Projection. Recently a EMT-244 digital reverberation unit has been purchased.

7. Microphones

IRCAM's collection of microphones includes the following types:

- Neumann U-87 electrostatic,
- Neumann SM-69 stereo electrostatic (2),
- Neumann QM-69 quad electrostatic (1),
- Sennheiser artificial head (1).

8. Portable Installations

The portable installation is intended for outside recordings; for outside concerts the assembly of a portable unit is planned for 1980. Principal portable equipment includes:

- a) Nagra 1/4 inch stereo tape deck type 4S,
- b) Stellavox 1/4 inch quadraphonic tape deck type SQ-8,
- c) Studer portable mixer, 8 inputs, 4 outputs.

Personnel permanent de l'IRCAM en 1978

Directeur : Pierre Boulez

Conseiller Scientifique : Max Mathews

Responsable artistique : Nicholas Snowman

Responsable des Relations Extérieures : Brigitte Marger

Responsable Administratif et Financier : Yves Galmot

Adjoint : Claude Minière

Responsable du Personnel : Chantal de Girardier

Département Ordinateur

Responsable : Jean-Claude Risset

Ingénieurs/chercheurs : Raymond Bara

Peter Eastty

James Lawson

Philippe Prévot

Jean-Louis Richer

Département Electroacoustique

Responsable : Luciano Berio

Ingénieurs/Chercheurs : Alain Chauveau

Giuseppe di Giugno

Jean Kott

Département Instruments et Voix

Responsable : Vinko Globokar

Ingénieur/Chercheur : René Caussé

Technicien : Michel Ducoureau

Département Diagonal

Responsable : Gerald Bennett

Ingénieurs/Chercheurs : James A. Moorer

Bennett Smith

David Wessel

Compositeur : Tod Machover

Ingénieur du son : Ben Bernfeld

Assistant : Didier Arditi

Responsable de l'Espace de Projection : Alexis Barsacq

Assistants : Jean-Louis Aichorn

Guy Costado

Techniciens : Jean-Paul Léglise

Didier Roncin

Département Pédagogique

Responsable : Michel Decoust

Coordination Technique

Responsable : Jean-Pierre Armand

Assistant : Georges Giscard

Bibliothèque

Responsable : Ivanka Stoianova

Guests invited to work at IRCAM during 1978

- Curtis ABBOTT (USA) - Development of a control language for the synthesis system built by Giuseppe di Giugno.
From September, 1978.
- John AMUEDO (USA) (College of Creative Studies, University of California, Santa Barbara) - Developed digital signal processing algorithms for extracting features of live musical instrument sounds in real time.
From September to November, 1978..
- Daniel ARFIB (France) (Acoustique Informatique Musique, LMA CNRS Marseille) Research on the program Music V: adaptation of new functions to linear distortion.
February, 1978.
- Vito ASTA (Italy) Collaboration with Giuseppe di Giugno on software for the 4C system. Aided Shmuel Bolozky on the project "Fast Speech".
From January to October, 1978.
- Arthur BENADE (USA) (Professor at Case Western Reserve University, Cleveland, Ohio) - Experimental and theoretical work leading to the design of a new flute.
June, 1978.
- Shmuel BOLOZKY (Israel) - Research project "Fast Speech".
From July to August, 1978.
- Michèle CASTELLENGO (France) (Assistant research fellow at the "Laboratoire mécanique/acoustique" of the University Paris 6) - Study of the perception of multiphonic sounds of the flute and of reed instruments.
DGRST Project from April to December, 1978.
- Conrad CUMMINGS (USA) (Center for Electronic and Computer Music, Brooklyn, NY) - Assisted in analysis of recorded instrumental sounds.
June, 1978.
- John CHOWNING (USA) (Professor of Music, Stanford University, Stanford, California) - Research on the control of sound spectra in direct digital synthesis.
From December, 1978.
- Jean-Luc DELATRE (France) - Programmer for the project "Study of speech analysis and synthesis".
DGRST Project from March, 1978.
- Jean DERECHAPT (France) (Dept. Mathématiques et Informatique CNAM).
Thesis on the simulation of sound reverberation on the computer.
From January to May, 1978.
- Robert DICK (USA) (Flutist - Fellow of the center for the creative and performing Arts SUNY at Buffalo USA) - Directed the project "A new flute".
DGRST project from July to December, 1978.
- David EHRESMAN (USA) (Psychologist and Data Analyst - Michigan State University, East Lansing, Michigan, USA). Installed ALSCAL, MORALS, PROFIT and related programs for exploratory data analysis. Assisted in development of sound editing program on the PDP-11 computers.
From June to July, 1978.

- Andrew GERZSO (Mexico) (Flutist) - Work on contact microphones. Collaborated with Pierre Boulez for Collège de France seminars, instructor for Session de Formation for composers.
From January to December, 1978.
- Thorsteinn HAUSSON (Iceland) (Composer) - Research on the use of inharmonic sounds for composition.
From November, 1978.
- Stanley HAYNES (Great Britain) (Composer) - Assisted in the realization of the composition "Arcus", by York Hoeller, for instruments and tape synthesized by computer, later documented this work.
From July to September and from November to December, 1978.
- York HOELLER (Germany) (Lecturer for Music Theory, Hochschule fuer Musik, Cologne) - Realization of the composition "Arcus" for instruments and tape synthesized by computer for the opening of the Espace de Projection.
From July to September, 1978.
- Mark KAHRS (USA) (Engineer, Center for Computer Research in Music and Acoustics - Artificial Intelligence Laboratory, Stanford University, California) - Preparation for the program NSYNTH for the manipulation and the visualization of acoustical data.
From June to August, 1978.
- Thomas KESSLER (Switzerland) (Composer - Professor of Composition and Theory of Music at the Academy of Music, Basel) - Research on contact microphones.
From August to October, 1978.
- Lev KOBLYAKOV (Israel) (University of Jerusalem) - Research project on musical analysis of serial works from 1950 to 1965.
From October, 1978.
- Denis LORRAIN (Canada) - Work on digital sound synthesis, digital sound transformation, and the development of algorithms for automatic composition. Analysis and re-creation of elements of "Inharmonique" by Jean-Claude Risset.
From June to September and from November, 1978.
- Istvan MATUZ (Hungary) (Flutist - Professor at the Academy of Music of Debrecen) - Research on the acoustics of the flute, especially multiphonics. From August to October, 1978.
- Mario MILANI (Italy) - Program development for the synthesizer projects 4A and 4C.
From January to June, 1978.
- Janos NEGYESY (Germany) (Violonist) - Documentation on the technique of the contemporary violin.
DGRST project from January to December, 1978.
- Alexandre PARODI (France) (Student at the Ecole Nationale Supérieure de Physique, Marseille) - Investigation of new processes for sound synthesis and processing.
From September to November, 1978.
- Xavier RODET (France) - Directed the project "Study of speech analysis and synthesis".
DGRST project from March, 1978.

Neil ROLNICK (USA) (Composer) - Documentation of all computer programs at IRCAM concerning the synthesis and analysis of sound. Musical and technical assistant for the realization of "Wellenspiele" by Balz Truempy.
From November, 1978.

Gideon ROSENGARTEN (Israel) (The Hebrew University of Jerusalem) - Research in musical analysis on the basis of information theory.
From January to April, 1978.

Philippe-Olivier ROUSSEAU (France) (Student at the Ecole Nationale Supérieure des Télécommunications) - Study on analysis and synthesis of instrumental sounds by linear prediction.
From March to June, 1978.

Maurice ROZENBERG (France) - Realization of programmable filters controlled by a micro-processor.
From January, 1978.

Klaus RUNZE (Germany) (Professor of Piano and Pedagogy, Hochschule fuer Musik, Cologne) - Catalogue and analysis of paintings realized by children in direct relation with specific musical tasks.
From May to October, 1978.

Johan SUNDBERG (Sweden) (Professor, Royal Institute of Technology, Stockholm) Research on the physiology and acoustics of singing, work on the synthesis of the singing voice.
From December, 1978.

Balz TRUEMPY (Switzerland) (Composer - Professor of Composition and Theory of Music, Academy of Music, Basel) - Realization of a composition commissioned by IRCAM for the opening of the Espace de Projection using the 4C Machine.
From May to September, 1978.

BIBLIOGRAPHY

- ABBOTT, Curtis - "A Software Approach to Interactive Processing of Musical Sound" (1978), Computer Music Journal, Vol. 2, Nr. 1.
- "Machine Tongues I, II, III" (1978), Computer Music Journal, Vol. 2, Nr. 1, 2, 3.
- AMUEDO, John - "Computers in Live Musical Performance" (IRCAM report).
- "An Information Processing System for Musical Performance" (IRCAM report).
- ARFIB, Daniel - "Digital synthesis of complex spectra by means of multiplication of distorted waves", preprint of the Convention of the Audio Engineering Society, Hamburg, February, 1978.
- ASTA, Vito - "Un modèle de machine musicale interagissant avec l'homme: caractérisation théorique et problèmes de réalisation" paper given at the UNESCO meeting on computer music, Aarhus, Denmark, August, 1978.
- "Tecniche di sintesi per la musica elettronica: la situazione attuale e i recenti sviluppi all'IRCAM", paper given at the Sixth Convention of the Italian Acoustical Society, Ivrea, October, 1978.
- BENNETT, Gerald - "Research at IRCAM in 1977" (IRCAM paper 1/78).
- BERNFELD, Ben - "Tétraphonie et stéréophonie à tête artificielle", Zéro VU, Nr. 5, mars 1978.
- BERNFELD, Ben, SMITH, Bennett - "Computer-aided Model of Stereophonic Systems" (IRCAM paper 15/78).
- BOLOZKY, Shmuel - "Report on Progress of Fast Speech Projects" (IRCAM report).
- CASTELLONGO, Michèle - "Etude Acoustique de l'Emission multiphonique au trombone à coulisses" (Rapport DGRST, IRCAM).
- CAUSSE, René - "A Programmable Device for Transforming Sounds in Real-time Under the Control of a Performer" (IRCAM paper 2/78).
- DECOUST, Michel, and ARFIB, Daniel - "Documentation pour l'oeuvre 'Interphone' par Michel Decoust, oeuvre réalisée à l'IRCAM en 1977" (IRCAM report).
- DELATRE, Jean-Luc - "Le système de traitement de signaux digitaux 'Junior'" (IRCAM paper to be published).
- DERECHAPT, Jean - "Etude de la réverbération sonore par simulation sur ordinateur". Mémoire présenté en vue d'obtenir le Diplôme d'Ingénieur I.I.E., soutenu le 21 juin 1978. (Directeur de recherche J.A. Moorer.)
- DICK, Robert - "La nouvelle flûte" (Rapport DGRST, IRCAM)
- GIUGNO, Giuseppe di, ALLES, Hal - "A One Card 64 Channel Digital Synthesizer" (IRCAM paper 4/78).
- GLOBOKAR, Vinko - "10 Exemples pratiques sur le rapport texte/musique", dans "La quinzaine littéraire", août 1978, numéro consacré à la musique nouvelle.

- "Réflexions sur l'improvisation" (IRCAM paper to be published).
- HAYNES, Stanley - "Rapport sur la réalisation technique de la pièce "Arcus" de York Hoeller" (IRCAM report).
- "Possibilités arithmétiques des passes 1 et 2 dans la nouvelle version de Music 5 à l'IRCAM" (IRCAM report).
- LORRAIN, Denis - "Analyse de la bande magnétique d'"Inharmonique" de Jean-Claude Risset" (IRCAM report).
- "Canons stochastiques" (IRCAM report).
- MATHEWS, Max and BENNETT, Gerald - "Real-Time Synthesizer Control" (IRCAM paper 5/78).
- MILANI, Mario, GRIPPE, Ragnar - "Acoustical Illusion" (IRCAM report).
- MOORER, James Anderson - "The Use of Linear Prediction of Speech for Computer Music Purposes", AES conference, Hamburg, February 1978 preprint (IRCAM paper 6/78).
- "About This Reverberation Business" (IRCAM paper 17/78).
- "Synthèse de la parole pour les buts musicaux" (NATO-ISI conference in June 1978).
- "How a Computer Makes Sound", (Computer Music Journal, Summer 1978)
- PARODI, Alexandre - "Synthèse additive par signaux périodiques quelconques" (Rapport de stage 1978).
- POLI, Giovanni di - "Musica, Programme de codage de la musique" (IRCAM paper 7/78).
- RICHER, Jean-Louis - "Music V - Manuel de référence" (IRCAM report).
- RISSET, Jean-Claude - "Musical Acoustics" in the Handbook of Perception, vol IV, (E.C. Carterette and M.P. Friedman, ed., Academic Press, 1978, pp 521-564).
- "The Musical Development of Digital Sound Techniques" (Invited Paper, UNESCO Meeting, Final Workshop in Computer Music, Aarhus, Denmark, August, 1978).
- "The Development of Digital Techniques: A Turning Point for Electronic Music" (IRCAM paper 9/78), Nutida Musik (Stockholm) 1979.
- "Paradoxes de hauteur" (IRCAM paper 10/78).
- "Hauteur et timbre des sons" (IRCAM paper 11/78), Bulletin d'Audiophonologie 8 (1978) nr. 3, pp 7-26.
- RODET, Xavier, DELATRE Jean-Luc - "Time-domain speech synthesis-by-rules using a flexible and fast signal management system" (IRCAM report), published in Proc. IEEE ICASSP, Washington, April, 1979.
- ROLNICK, Neil - "A Composer's Notes on the Development and Implementation of Software for a Digital Synthesizer" (IRCAM paper 18/78).

ROZENBERG, Maurice - "A Digitally Programmable Filter" (IRCAM paper 16/78).

- "Microcomputer-controlled sound processing using Walsh Functions", Computer Music Journal, Vol. 3, no. 1).

SMITH, Bennett, EHRESMAN, David - "A Computer-Controlled, Display-Oriented Sound Editor" (IRCAM report).

STOIANOVA, Ivanka - Geste, texte, musique, (Paris, L'Union Générale d'Editions, 10/18, Série Esthétique, Paris, 1978.

- "Musique répétitive" (Epistémé, 10, vol 4, II/1978, Tokyo, pp. 8-20).
- "La musique - utopie d'après Bloch et la musique occidentale contemporaine", (Le discours utopique, Union Générale des Editions, 10/18, 1978, pp. 153-160).

SUNDBERG, Johan - "Le chant comme objet de recherche scientifique" (conférence donnée à Lyon en Janvier 1978).

WESSEL, David - "Low Dimensional Control of Musical Timbre" (IRCAM paper 12/78). Also published under the same title as Audio Engineering Society Preprint No. 1337 (D-4) and presented at the 59th Convention of the AES March, 1978, Hamburg, Germany.

WESSEL, David, EHRESMAN, David - "Perception of Timbral Analogies" (IRCAM paper 13/78).

WESSEL, David, GREY, John - "Conceptual Structures for the Representation of Musical Material" (IRCAM paper 14/78).

WESSEL, David, SMITH, Bennett, EHRESMAN, David - "Psychoacoustic Experimentation as a Prelude to Musical Composition" (Presented at the 96th Convention of the Acoustical Society of America, Honolulu, Hawaii, December, 1978). Abstract published in The Journal of the Acoustical Society of America, Vol 64, Supplement No. 1, Fall, 1978, p S170.

WESSEL, David, SMITH, Bennett - "Psychoacoustic Aids for the Musician's Exploration of New Material" Presented at the 1977 International Conference on Computer Music, University of California at San Diego. A written version is distributed by the Computer Music Journal.

Records published in 1978 of works realized at IRCAM.

RISSET Jean-Claude - Mutations, Dialogues, Inharmoniques, Deux Moments Newtoniens - I. Jarsky, soliste, direction Michel Decoust, Michel Philippot - INA-GRM AM 564.09 (Label Musique Française d'Aujourd'hui.)

- "Inharmonic Soundscapes", New Directions in Music (Tulsa Studios, 1978).

David Wessel - "Antony", New Directions in Music, (Tulsa Studios, 1978).

Compositions realized at IRCAM during 1978.

BIRTWISTLE Harrison - Elements for "Oresteia", in collaboration with the National Theatre of London, realized on the PDP10 and on PDP11/03 equipped with the digital synthesizer "4B". Assistants: Jonty Harrison, Neil Rolnick and Philippe Prévot.

HOELLER York - "Arcus" for instruments and quadraphonic magnetic tape (IRCAM commission). Tape synthesized on the PDP10. Assistant: Stanley Haynes et David Wessel. Duration 13'.

MOORER James Anderson - "Lions are growing", tape synthesized by computer from modified human voice. Duration 2'40.

- New realization of the composition "Perfect Days" (1977, for transformed voice and flute). Duration 1'30.

RISSET Jean-Claude - "Mirages" for 16 instruments and magnetic tape (commissioned by the Donaueschingen Festival). Quadraphonic tape synthesized on the PDP10. Duration 23'.

ROLNICK Neil - "Wondrous Love" for trombone and quadraphonic magnetic tape. Tape synthesized on the PDP10. Duration 10'.

TRUEMPY Balz - "Wellenspiele" for instruments and digital synthesizer (commissioned by IRCAM). The 4C system was used in the performance of the work. Assistant: Neil Rolnick. Duration 20'.

Members of IRCAM presented papers at the following conferences:

- 59th Convention of the Audio Engineering Society, Hamburg, March, 1978.
- 9èmes Journées d'Etude sur la Parole du Groupe Communication Parlée du G.A.L.F., Lannion, France, May - June, 1978.
- UNESCO meeting on Computer Music, Aarhus, Denmark, August, 1978.
- Third International Computer Music Conference, Chicago, November, 1978.
- 96th Convention of the Acoustical Society of America, Honolulu, Hawaii, December, 1978.
- Sixth Convention of the Italian Acoustical Association, Ivrea, October, 1978.
- Club Informatique de l'Université d'Orsay.
- Club Informatique de l'Université de Nancy.
- Maison de la Culture de Grenoble (Exposition "Informatique et vie quotidienne").
- Studio Heinrich Strobel, Freiburg, Germany.

- Aspekte, Salzburg.
- Seminar Klaus Huber, Hochschule fuer Musik, Freiburg, Germany.
- 1978 IEEE International Conference on Acoustics, Speech and Signal Processing, Tulsa, Oklahoma, USA, May - June, 1978.
- Computer Music Symposium, World Music Days, International Society for Contemporary Music, Stockholm, May 1978.
- Colloque sur l'Utopie, Centre Thomas More, Lyon, April 1978.
- Colloque "Création musicale et futur", Universidad Nacional Autonoma, Mexico City, December, 1978.