TASK QUARTERLY 3 No 2 (1999), 159–170

COMPUTER APPLICATIONS IN MUSIC — A SURVEY

WŁODZIMIERZ KOTOŃSKI

Fryderyk Chopin Academy of Music in Warsaw,
Okólnik 2, 00–368 Warszawa, Poland

Abstract: Two main applications of computers in music - sound synthesis and simulation of the compositional process are presented in this paper. Other applications of computers, such as digital sound recording or editing and mixing of sound tracks are also objects of discussion.

Keywords: digital sound synthesis, digital registration of sound, digital sound processing, sampling and samplers, music programs, digital music instruments, simulation of compositional process.

1. Introduction

Computer application in art is quite common nowadays in such fields as, for example, fine arts, film animation, architectural design, and last, but not least, in music. As this specific application of computers is rather not popular outside the specialist circles, I would like to demonstrate it to all the computing machines users, who would be interested in the subject.

The term „computer music", which can be met quite often, has many interpretations and always causes controversies. Is it music created by a computer itself, or music created by a composer, yet the sound is realized with the use of a computer, or, finally, is it music created in a traditional way, whose creation was performed through a computer as a tool. All these meanings are used at present, although maybe not in that simple and univocal way, and many composers and performers use modern technology, sometimes not even realizing that they are dealing with highly specialized computer systems. The term most often used among the creators and performers of music is: *digital* instruments and devices. In most of them a computer as such, with its alphanumeric keyboard, screen, or discs drive is not needed at all. Yet we have to deal here with two most important parts of a computer — *processor* and *memory*, and also with *programs* which are often hidden away from the user. I mean here not only those simple devices which are commonly called „keyboards", but also more complicated digital synthesizers, samplers, or harmonizers, often also equipped with a keyboard similar to that of a piano. To this is added a whole range of processing and memorizing devices, such as filters and correctors, reverberation units, digital acoustic effects, sequencers, electronic drums, and such highly specialized instruments, as „Clavinova", and finally a whole range of MIDI instruments with an acoustic piano operated from a sequencer.

Already 90% of today's pop music are computerized, taking into consideration the fact that most of the instruments used nowadays are digital ones. Very often the computer itself appears on stage, and even more often it can be seen in the recording studio. In artistic music we can meet digital electronics more and more often at modern music concerts and in electronic music production studios that are often called computer music studios because of the kind of music produced. Composers as well buy personal computers with ready–made music programs, which allow them to use their computers as notebooks or secretaries and copyists. The more advanced in the field of computer science ones try to use their computers to simulate the creative processes of composing music by machines. They also try to interact with the program, or, even going further, in the concert situations they try to use the interaction between the performer and a computer. This is the situation for today.

Let's now try and introduce some order into our divagations. Let's leave the term „computer music" for a moment and discuss in detail the role of computers and digital equipment in music. This role changes depending on the needs, equipment and programs that the user has at his disposal.

Computer applications in music can be divided into:

a) digital sound synthesis;

b) digital sound processing;

c) simulation of compositional processes;

d) interactions during musical production;

e) electronic (digital) sound editing;

f) music application in teaching (e. g. ear training);

g) applications that are not necessarily connected with music (e. g. organizing a sound library).

2. Digital Sound Synthesis

Digital sound synthesis is a very important field in the computer application in music. We deal here with three types of actions: resynthesis, direct synthesis and mediate synthesis.

2.1 Sound resynthesis

It consists in a natural sound analysis and then creating with digital equipment a sound similar to the original one as much as possible, or creating a different sound, yet basing on a close representation of the natural sound. When we talk about a sound we should understand the term as either a separate sound or a sequence of sounds consisting of several similar or different sound units.

To understand resynthesis fully it is necessary to get acquainted with the rules of digital acoustic signal registration, which is often (and mistakenly) called „digital recording". An acoustic signal travelling in the air as a variable pressure wave is changed into an electric signal by a microphone and an amplifier. This is an analogue signal. The current value changes, occurring most often as changeable volume, are not identical, but analogous to the changes of the acoustic signal. An electric analogue signal is continuous, like its prototype acoustic signal. Its time chart can be presented as a continuous function. Such a signal can be recorded as a changeable groove on a record, or a changeable magnetization of a tape. These are analogue recordings.

Computers and all the devices cooperating with them work in a discontinuous system. We should remember that they were at first counting machines. Only later it became clear that numbers can also function as symbolic phrases and that it is not only possible to use them for counting but also for logical operations. How is it possible then to record an acoustic signal curve, which is analogous hence continuous, by the use of digits? A continuous curve is changed in an analog–to–digital converter into a row of digits possessing certain parameters (time flow, deviation from constant, e.g. zero). The points whose distance from x axis is marked by the momentary amplitude value, and which usually appear in equal time intervals, are called samples. The distance between samples is called sampling rate, and is expressed by the number of samples per second, that is in Hz. In the experiments it was possible to obtain an optimum sampling density, which allows to create the impression of continuity and a proper, not deformed „picture" of sound. It is between 30–50 thousand measure points per second. In the case of sound a higher density (than 50Hz) is not economical, as it no longer gives any perceived improvement, and a lower one (than 30Hz) causes a perceptible deterioration of sound quality. In digital recordings (CDs) and in most DAT appliances, so called digital recorders, a standard of 44.1 kHz has been assumed (exceptionally 48kHz for some DATs). The quality of production is not only influenced by the density of sampling, but also by the precision of measurement, called quantization. In decimal numbers this precision is stated by the number of places after the decimal point, and in binary systems by the bit number. As for today a measurement expressed by a 16–bit number is good enough for sound reproduction, although some computers work in the 32–bits system. A so–called digit signal, which is an extremely long row of binary numbers sting the amplitudes of individual samples is registered on a magnetic tape, CD record, or on hard disc. For signal reproduction one more half is needed. That is a digital–analogue converter, in which a digital signal is transformed into variable voltage and a smoothing filter converts the received stepped voltage into changeable voltage in a continuous way. The signal smoothed in that way is then directed onto an amplifier and speakers, to finally become an acoustic signal very close to the registered one. This resynthesis, which really is nothing more than just registration and reproduction, is extremely important in sound recording industry, radio, TV, or film recordings. The quality of reproductions is much higher than the one used in analog recordings. The kind of deformations is less painful, and their level in digital recordings is much lower. The distance between the own noise and useful signal is so big (100 dB) that it can be neglected in assessment.

A composer, however, is not as much interested in a possibly close sound reproduction, as in creating his own sound units and the possibility of operating them in his own way. Hence he will use the fact that between the two halves–the change of electric signal into a digital one and a digital one into analog one–there is this „middle" in the form of digital signal registered on a magnetic tape or hard disc. This signal can be operated on in many different ways. We can change its pitch, dynamics and time. The signal can be filtered, cut into pieces and then spliced together, everything, of course, within numerical operations. Through these actions a composer can change the initial signal drastically, preserving at the same time some of its features and properties. At present highly specialized devices called *samplers* are used for resynthesis. They allow to digitally register any sound or sequence of sounds (for example some words), called here a sample, and then to reproduce it on various pitches, with different dynamics, and in different times by the use of a musical keyboard or controlling signals from a computer program. The simplest way of using a sampler, most often used by pop musicians, is the use of sound samples provided by a producer (so–called *presets*) which are digitally registered, isolated instrument sounds or sound effects. Musicians with higher ambitions, however, try to create their own sounds either through transformation of presets, or by recording their own samples and their transformation. Here a computer appears to be very useful. A special program for analysis and transformation, coordinated with a given type of sampler allows operating the sound easily. Although it only controls the sampler modules, it makes it possible to actually see the changes introduced on a screen, and to control them acoustically by the use of a speaker. Recently integrated systems for registration, transformation and editing the sound have appeared. They are all included in one program (e.g. PRO TOOLS). Both the initial registration and the final effect are placed in the computer's memory on hard disc, without the use of outside equipment.

2.2 Direct synthesis

Direct synthesis consists in rejecting the first half from resynthesis, and starting the work from the middle. The creator starts with creating his own sound in the form of certain digit systems. He will not, of course, put the sound together sample by sample. He is going to use trigonometric tables and linear tables, adding, multiplying, logical operations. He will prepare a set of algorithms and procedures. He will also construct a program, which will allow placing sound units in the proper time points, with a proper pitch, duration time and dynamics. The whole course of signal and data for the next samples are automatically calculated by the computer. To make the synthesis properly it is necessary to posses the knowledge of acoustics and previous experience with resynthesis.

Direct synthesis is historically the oldest one. It was performed at the end of the 60's by Max M. Mathiews and Jean Claude Risset in the Bell's Laboratories in New Jersey, together with their work on instrumental and speech sounds analysis. Mathiews created the first special computer synthesis program, better known from its later version MUSIC V. It was used and upgraded for several years afterwards. Nowadays programs of that type are no longer used because they are too slow and complicated. The breakthrough came around the year 1980, together with the introduction to the market microcomputers (IBM PC, Apple Macintosh) and digital synthesizers, with the famous YAMAHA DX7. The crucial element of the latter were digital generators included in the hardware, so not requiring programming typical functions, e.g. sinusoidal signals. Here we are entering mediate synthesis. Yet the direct synthesis programs are still used in big laboratories and institutions (e.g. CCRMA in Stanford, IRCAM in Paris), as they allow to create sounds impossible to obtain in any other way, although the process is a lengthy and complicated one. There also have appeared commercial direct synthesis programs for small personal computers.

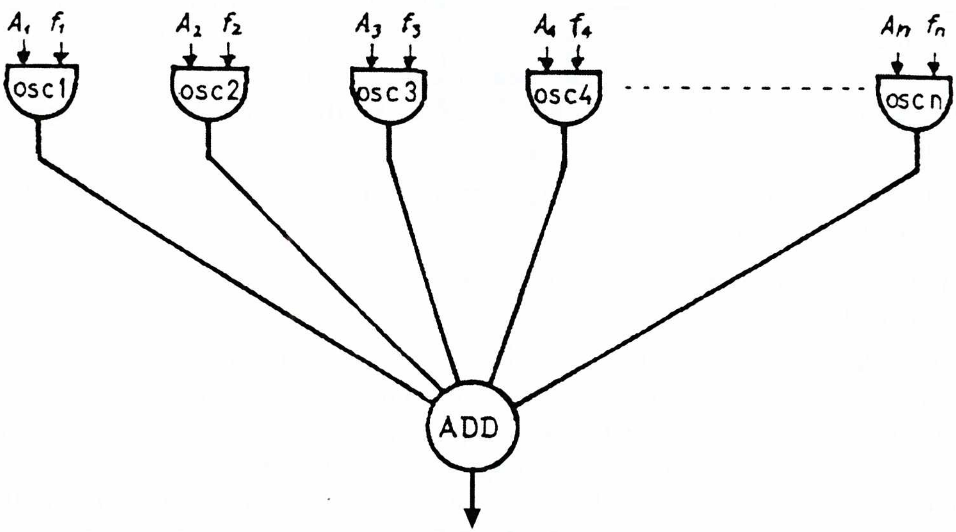

Figure 1. *Additive Synthesis:* $A_1, A_2, ..., A_n$ *— oscilators amplitude,* $f_1, f_2, ..., f_n$, *— oscilators frequency, ADD — Adder*

In the field of direct synthesis we can differentiate among: additive synthesis (putting sounds together by adding simple tones), subtractive synthesis (by filtering sounds which are rich in harmonic or non–harmonic components from frequency bands), synthesis performed by frequency or other kinds of modulation, formant synthesis and synthesis basing on artificial intelligence in the form of neural net, allowing to copy the ways of inducing and resonating sounds, and not their effects as ready acoustic phenomena.

2.3 Mediate synthesis

Mediate synthesis is the most common way of creating synthetic sounds. It bases on the digital generators mentioned above. A synthesis of this kind can also be performed without a computer, yet the digital synthesizers by which it is done are in reality highly specialized computers with musical abilities: through the keyboard, buttons and switches. Instead of a screen they have small windows which present names and digital data. Synthesizers work using either additive synthesis or, most often, different types of signal modulation. The most important one has become frequency modulation (FM), on which YAMAHA DX and SY digital synthesizers were based.

3. MIDI System

Digital synthesizers would remain just specific musical instruments (they can be played like piano or organs), if it were not for an idea of a few companies producing digital equipment (Sequential Circuit, Roland, Yamaha among them). The initial idea was to operate several synthesizers from one keyboard, and was called MIDI (Musical Instruments Digital Interface). The MIDI system appeared to be a very universal one, ideal for transferring data among synthesizers and other digital devices (of the same or different firms), and between digital devices and a computer. This system is quite complicated and it is difficult to describe it in a few sentences. I will only try to describe its function. A MIDI–converter at the output of each device codes data concerning:

a) the keyboard status–which key is pressed, at what speed it has been attacked, and how long it was pressed;

b) the condition of devices of changeable parameters and the connections between modules.

The b) data are transferred in groups out of the musical time, as so–called bulk data. They serve to specify the type of „instrument" and other data concerning, generally speaking, the sound characteristics. The a) data are transferred within the musical time, together with the pulses of a generator called „clock". They concern the condition of the key board, or rather „notes": they can be stored in the memory of a so–called sequencer. Sometimes it is a separate device, yet most often it is a program module of a computer. This allows to modify later the sequences recorded in the MIDI signal (or even separate notes) by cutting, replacing, copying, layering, etc. All this takes place in the computer program, outside the synthesizing part. If we add to all this the possibility of transferring the MIDI data to the graphic program printing notes in musical notation, it will appear that the MIDI system application has exceeded by far its initial purpose. Since its introduction in 1981, within the next 10 years it has become a universal conversation system, joining devices of different companies, a system without which it would be difficult to imagine modern electronic music (digital).

Now–some more about synthesis by the use of frequency modulation. Frequency modulation has been known since the beginnings of radio broadcasting (for transmitting signals along ultra–short waves). In the times of manual work on analog devices it was used for creating new sounds. Its applicability for creating harmonic sounds was re–discovered by John Chowning from Stanford University, California, in1968. With the precision and stability of digital generators, checked in laboratory conditions at that time, it appeared that by the use of frequency modulation of two sinusoidal signals of different dynamic envelopes, when both signals are within acoustic frequencies, it is possible to create complex harmonic and inharmonic spectra of different time placement. In other words it is possible to create complex instrumental sounds of qualities changing together with the sound development. This allowed creating synthetic sounds of complexity similar to the natural sounds, and also for quite close imitation of acoustic instruments (violin, clarinet, trumpet, etc).

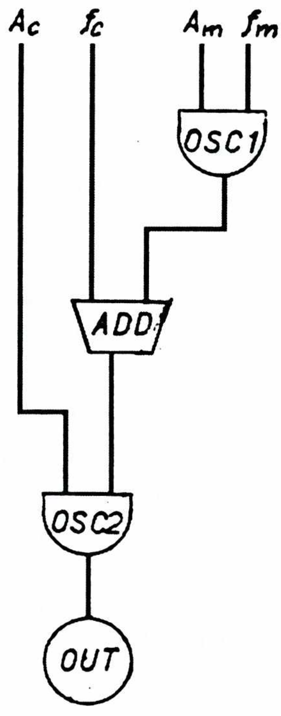

Figure 2. *Frequency modulation.A_c — carrier amplitude, A_m — modulation amplitude, f_c — carrier frequency, f_m — modulation frequency*

The modulation was a more precise one if several generators were used, modulating the frequency of one, or in a sequence–each one modulating the frequency of the next one (so–called cascade modulation). Chowning sold his idea to the Yamaha Company, which introduced this system of modulation into all their digital synthesizers. Other firms, such as ROLAND or AKAI use mixed systems: additive synthesis (including generating rectangular and saw tooth shaped signals), ring modulation (old because used already in the first Stockhausen's electronic trials), and sampling, that is ready fragments of a digital recording. All that is, of course, coupled with the MIDI system.

4. Other Computer Applications in Music

Sound synthesis seems to become the most important one among various computer applications in music. It has found its continuation in specialized digital devices, constituting a part of a computer each, and connected with special input–output devices.

Apart from synthesis computers in music have many totally different applications. This has been really growing in the last few years. It is connected with the miniaturization and greater availability on one hand, and commercial programs development on the other. Let's remind here that in the 70's, for example, so only some 20 years ago, computers were only available in rich university centers. Only those who possessed the difficult art of programming could undertake to work with them. In the field of music these were either people of double mathematical and musical education (like John Chowning), mathematicians or physicians yet musical amateurs (like Max Mathiews). Only the introduction of relatively cheap yet fast and capacious PC microcomputers, which were available to less wealthy institutions and private users, together with the appearance on the market of many commercial programs made the computers more popular, and especially those used for musical purposes. The first programs could also be used by people who knew nothing in the field of computer sciences. This is the history of the last several years. The first Apple microcomputer appeared in 1981, and the firs music programs in 1985. The note printing programs appeared only around 1987, and are still being changed and upgraded.

Computer programs (those ready ones, comfortable for the user) can be used:

a) to co–operate with a composer (also arranger, improvising musician, etc.). These are sequential programs, basing on the MIDI system, with the possibility of introducing different transformations, up to using random actions for deciding some of the parameters (e.g. pitch, dynamics, time). These programs are also useful for composing traditional instrumental music (e.g. NOTATOR, CUBASE, SCORE PERFECT);

b) for printing music in common musical notation; these are graphic programs (in dots or vector graphics), which make it possible to prepare professional score pages to be photocopied later. Some of the programs are typically publishing ones (SCORE II, FINALE), others combine the advantages of a sequential program with the printing one (NOTATOR, ESCORT, CUBASE). In all of them it is possible to introduce notes, musical symbols and texts from the MIDI keyboard or with a mouse;

c) as interaction programs. These are the programs allowing the co–operation between a composer or performer with digital equipment in such a way, that some acoustic or physical actions (detected for example by a photo–cell) introduce respective changes in the signals generated by the synthesizer, start some fragments or influence the speed of reproduction;

d) as programs for sound analysis, presenting the signal charts as signal curves or spectra by the use of a short Fourier transform;

e) as programs co–operating with a given synthesizer or sampler, allowing to control the parameters of their modules, often connected with analyzing programs (e.g. C–LAB's X–ALIZER for YAMAHA DX synthesizers);

f) as programs for analyzing musical work. They are a bit more complicated and useful only for musicologists and music theoreticians;

g) as programs supporting ear training, more and more popular in the world;

h) as programs simulating creative composing processes. In this group there are only few commercial programs basing on random numbers generators (e.g. M., KANDINSKY). More often we deal with highly specialized programs, which can be found in huge science centers, such as IRCAM in Paris, CCRMA in Stanford, M.I.T. in Cambridge, Mass., or EMS in Stockholm. They are not accessible by home computers, but require very high speed and huge internal memory–the minimum being the NEXT computer. So work with these programs is not available to the beginners, as it requires advanced programming knowledge (e.g. C language). The effects are promising, however. It is not only about creating aleatoric structures, which in fact are the easiest ones to create, but, for example, generating musical structures in a particular historical style, or in accordance with a definite composing technique. Such structures, generated by a computer itself can be then accepted, refused or modified by a composer. In this case it is important to specify the criteria properly and economically, and to formulate precise rules according to which the structures should work. Because such rules are usually written in the form of algorithms, this way of creating music with the help of a computer is often called algorithmic composing or algorithmic music. A novelty are the recent experiments with artificial intelligence in a system called neural network, in which the programs are „taught" the model structures introduced by the user (melodies, harmonic sequences, typical rhythmical groups). This field of investigations, called connectionism, is getting more and more fashionable in the leading scientific centers, not only with relation to music. The idea is that instead of introducing into the program detailed rules leading to a required result, we only state the result and rely on the computer to do the job. Here we have got elements transferring with a precise weight from one level to another data, or their sum, which have to state the output values (see Figure 3). In the case of music the research is directed onto creating useful perceptive cognitive models, which could be used as tools for automatic creation of new compositions.
Generally we can say that an algorithmic composition consists either in deciding about detailed rules and using random actions to generate successive melody sounds, or in introducing into a recursive model of a neural network a sequence of examples. They will then be absorbed by the system (as a system

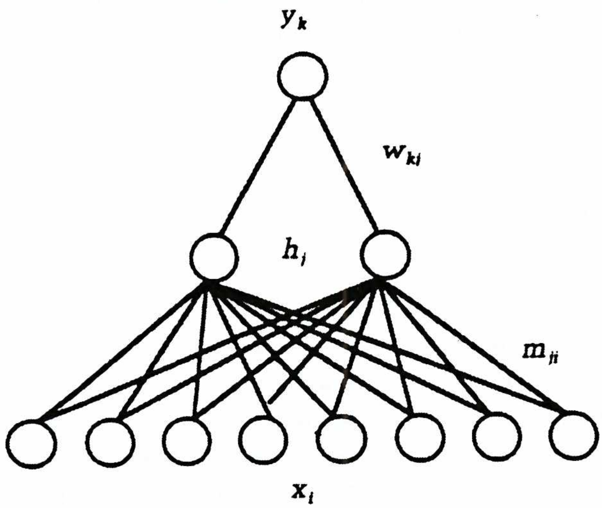

Figure 3. *Neural network. A 3–layered, one–direction network x_i — inputs, h_j — hidden units, y_k — output, m_{ij} and w_{kj} — weights*

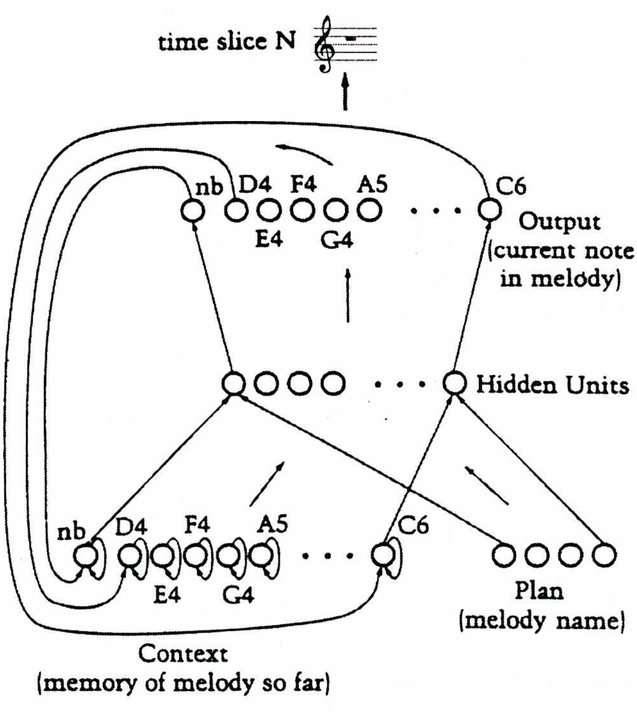

Figure 4. *Sequencial network design used for compositional purposes, the current musical representation requires note–begin (nb) and pitch units (D4–C6) for both output and context. Context units have also self–feedback connections (after Todd Peter M.,* A Connectionist Approach to Alghorithmoc Composition, *"Computer Music Journal", 13/4, 1989. © Massachussets Institute of Technology)*

of weight in passing from one layer of the network to another). As a result a general scheme of the process will be preserved, which will allow to generate for example a new melody similar to the model ones;

i) As programs used to register sound directly on hard disc, ignoring magnetic or laser devices. Such programs make it possible to put together, transform, and layer many registered sound sequences. At present they are used as standard programs for assembling and treatment of instrumental and vocal recordings in radio, sound recording, film, etc. They are also extremely important in working on multi–layer electronic music. The final effect is usually copied onto a DAT cassette or a CD. They are not really useful to private users because of their cost and equipment requirements. Most institutional electronic music studios, however, install such programs, which replace the whole digital music studios or only highly developed mixers' tables;

We could go on listing many more computer applications in music and musicology. We are not interested in a full catalogue here, however. This work is an attempt to review the problems which are the most interesting for musicians today (and not only composers).

And what is the condition of computer music in Poland? The main driving force of development in this field are digital and/or computer music studios. The oldest one is the Experimental Studio of Polish Radio in Warsaw, created in 1957. Then there are studios formed by music academies in Cracow, Warsaw, Wroclaw and Lodz. The Polish Radio Experimental Studio from the beginning was directed onto music production. Composers created their concert music on tape solo, an instrument, or for instruments/voice and tape. At the beginning they only worked in analog technique, as this was the only one available to them. Since the end of the 70's they have been switching to digital equipment. The advantage of working at the PRES was the fact that it had good technical devices, initially basing on the equipment generally used in the radio (generators, filters, stationary recorders). Later specialist equipment was developed, unwillingly financed by the directors of Polish Radio who sponsored the Studio. In this way computers appeared in the Studio. First these were Macintosh Apple II, later PC, and finally Macintosh Power, with a 4Gbyte disc. Also peripheral devices were added, such as sequencers, digital synthesizers, samplers, together with digital transforming devices, such as harmonizers, reverberators, etc. Also composers were sponsored through commissions, free access to equipment, and free assistance from the technical personnel of the studio. In the years 1959–1998 in the PRES over 100 compositions were created, some 20 of them partly or totally in the digital technology.

The studios affiliated with musical academies are primarily didactic units. They teach the students of composition, theory and sound engineering (the latter ones in Warsaw only) on the basis of work with synthetic sound, digital music recording systems, with digital transformation and digital sound assembly. Apart from taking part in lectures the students are also obliged to work practically at rehearsals and recordings, and first of all to compose music in the digital technology. Some of the studios also have scientific research centers on a small scale. In the studio of Musical Academy in Warsaw, for example, work connected with ordering and classification of all sound phenomena has been developed for several years now. The phenomena are classified according to their morphological properties, independently of their sources (continuation of P. Schaeffer's job). Also research on formalizing typical natural sound transformations is performed there. In Cracow academy the work on application of artificial intelligence in creating musical structures is being performed, together with work on the implementation of computers in ear training. Because of lack of proper equipment a lot of interesting work has been done by Polish composers and young scientists in big centers abroad. For example in excellently equipped E.M.S. studios in Stockholm they were working on various methods of direct and mediate synthesis, formant synthesis for simulating vocal sounds (co–operating with CHANT program). Another example can be CCRMA studios in Stanford, Cal. And IRCAM in Paris, where the main field of interest was simulating composing processes by computer systems. The result was creating a few interesting algorithmic compositions.

At the end it would be interesting to mention the contribution of Polish pop musicians and composers into the practical application of computers and digital devices. In the last years they have created a number of private, quite well equipped, production and recording studios, basing, of course, on digital devices and computers.

References:

[1] Chowning John M., *The Synthesis of the Complex Audio Spectra by Means of the Frequency Modulation*, " Journal of the Audio–Engineering Society" no 1, 1977, (Polish: *Synteza złożonych widm akustycznych przy pomocy modulacji częstotliwości* trans. W.Kotoński), "Zeszyt Naukowy Nr 31, Akademia Muzyczna w Warszawie", 1995

[2] Dolson Mark, *Machine Tongue XII, Neural Networks*, "Computer Music Journal" Vol. 13, no 3, 1989

[3] Mathews Max V., *The Technology of Computer Music* (Co–authors: J.E.Miller, F.R.Moore, J.R.Pierce, J.–C.Risset), Cambridge , Mass.– London, England 1969 (The MIT Press)

[4] Moorer James A., *Signal Processing Aspects of Computer Music–a survey*, "Computer Music Journal" Vol 1, no 1, 1977

[5] De Poli Giovanni, *A Tutorial on Digital Sound Synthesis Techniques*, "Computer Music Journal" Vol 7, no 4, 1983 (Polish: *Przewodnik po technikach syntezy cyfrowej* (trans. E.Sielicki), "Zeszyt Naukowy Nr 31, Akademia Muzyczna w Warszawie, 1995)

[6] Todd Peter M., *A Connectionist Approach to Alghorithmoc Composition*, "Computer Music Journal", 13/4, 1989

[7] Widmer Gerhard, *Qualitative Perception Modeling and Intelligent Music Learning*, "Computer Music Journal" 16/2, 1992