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ELECTRONIC MUSIC REPORTS

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The contents were as follows:

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The Electronic Music Studio at Stockholm, its Development and Construction

KNUT WIGGEN

ORGANISATION

The Electronic Music Studio in Stockholm (EMS) is run as a foundation, the most interested parties being the Ministry of Education and Swedish Radio. The foundation's board consists of representatives from various musical and closely allied institutions, and the chairman is appointed by the government.

The executive staff is made up of a studio director, an administrator, one secretary, two technical engineers, and three technicians; in addition there are two programmers and a producer who work part-time.

The running of the studio is financed in more or less equal parts by the state on the one hand and by Swedish Radio and other institutions on the other. The resources finance both the administrative and technical costs and the teaching courses. However, EMS receives no funds either for its own artistic work or for research. The compositions that are produced therefore express the musical wishes of individual composers and of the institutions that commission the works, not the daily problems of electronic music that the executive staff experience.

HISTORY

The composer Karl-Birger Blomdahl became Head of Swedish Radio's Music Department in 1964, and it was he who provided the starting point for the construction of the Electronic Music Studio in Stockholm, which was to be built by Swedish Radio. It was his ambition that the new studio should create technical bases in order to take electronic music a step further in its development. He did not want to copy any existing studio. It was a clear signal for the use of advanced technical solutions.

After Blomdahl's illness and death in 1968, Swedish Radio felt that it should no longer make use of the studio alone, but wished to share it with others. On the basis of this decision, it was decided to form the foundation as described above.

At the beginning of 1964, the requirements which were to form the basis of the construction were laid down. In the preparation of the studio's musical obligations, great consideration was given to current teaching conditions in order to prevent the studio from becoming a concern for only a small group of composers, because of the far-too sophisticated operating requirements.

The following were the requirements of the basic construction:

1. The method of production had to be wholly automatic, because any other method that required working on full-time over a longer period would exclude to all intents and purposes all non-established composers, and because the studio had to be able to accept commissions from radio, TV, films, theatre etc., and provide quality products in a short time.

2. The studio control system had to be so simple to operate, that virtually no composer would be prevented from using it because of the difficulties of operating it.

3. The construction had to have sound quality higher than what was usual in electronic music circles.

4. The studio, from a computer viewpoint, had to be near enough a "real time" construction; i.e. the sound had to sound at the same instant as the composer set the controls.

5. The development of computer music had shown that it was possible to compose music in a radical new way, wherein the composer writes a system of rules which control the computer's choice of the composition's individual tones. The studio had to anticipate the expected large expansion in computer music by also establishing a problem-free technical and musical link to external computers.

6. The studio had to be suitable to receive extensions.

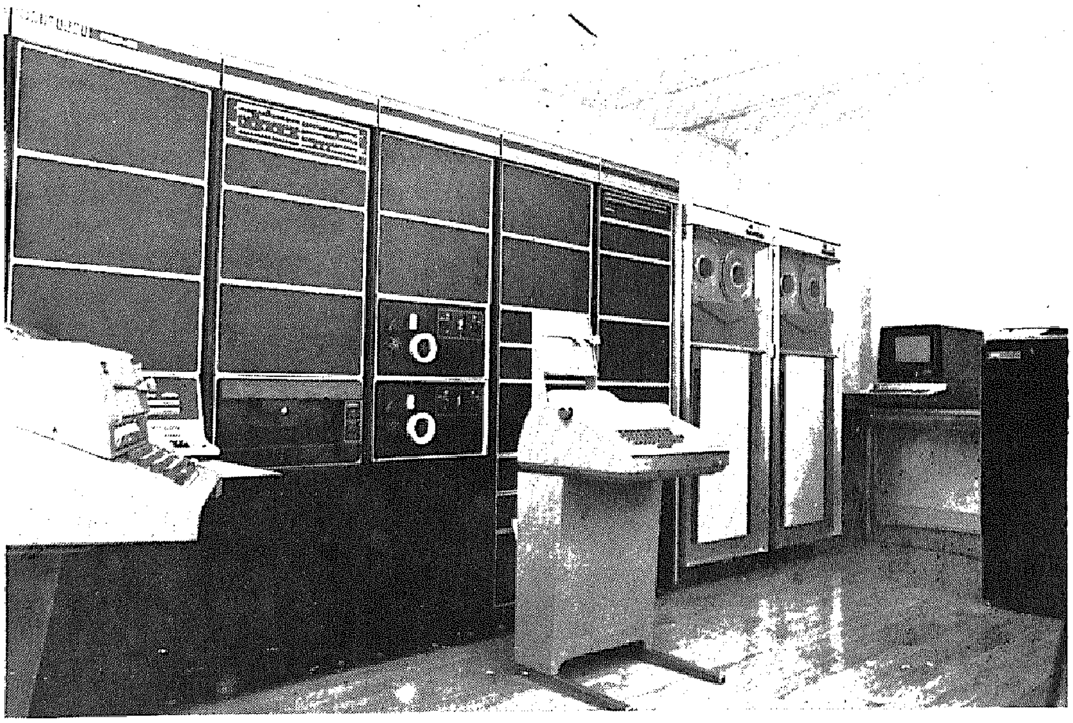
7. The studio had to fulfil scientific requirements both for performance and reliability.

TECHNICAL PRINCIPLES

The technical solution that we then chose was to allow the computer to control an apparatus which could both produce and re-work sounds. It would not be necessary for the computer to do the time-consuming work of calculating from the sound, and composers could work with the apparatus as they were used to, and avoid choosing between learning to handle the computer to a high degree or abstaining from musical experiment. The advantages were many, but so were the difficulties – which consisted of constructing the apparatus which the computer would control.

Such an apparatus scarcely existed on the market. Certainly pieces of apparatus that could be controlled by analog signals had begun to be built in the USA, but we considered the result unsatisfactory as far as the question of accuracy was concerned. With the resources that we had at our disposal, we elected to tackle immediately the difficult problem of controlling the apparatus by digital signals.

By autumn, 1966, the internal digital control system and most of the sound producing apparatus were operational. Two years later (1968), we could control the



EMS computer PDP 15/40

studio by means of an external computer. In the spring of 1969, EMS received money to order a computer which was supplied in different batches between June and November 1970. Work on the basic program, EMS-1, was begun in the autumn of 1969, and finished in the spring of 1972.

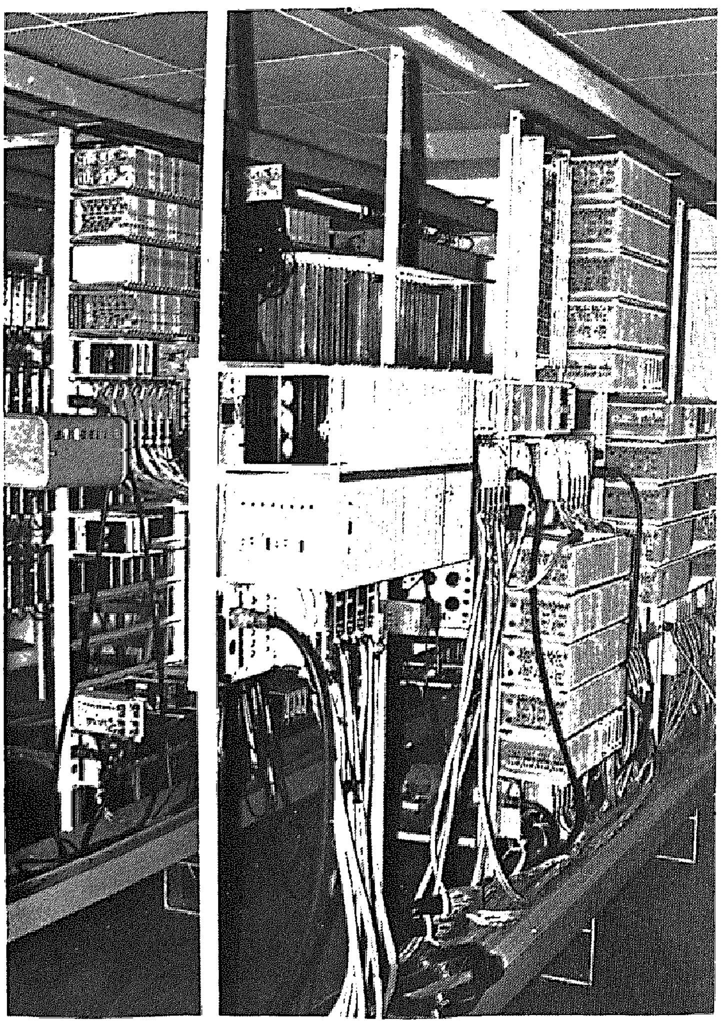
TECHNICAL EQUIPMENT

The core of EMS' apparatus is a computer of the type PDP 15/40 with the following specification.

1. One 32,000 word core-memory with an access time of 0.8 microseconds, and word length of eighteen bits.
2. Four disc-memories, each with a memory scope of 256,000 words.
3. Two fast digital tape recorders with a transfer rate of 50,000 words/sec.
4. Four slow DEC-tape units each with a capacity for 144,000 words.
5. One line printer, 300 lines/min.
6. One paper tape reader and one paper punch.
7. One clock of 250 kHz, that can interrupt the computer.
8. Two on-line type-writers that can control the two simultaneous computer programs.
9. One graphic and alphanumerical keyboard display terminal that controls the computer from the control console.

The computer controls the sound producing and sound processing apparatus. All of these apparatuses are digital controlled.
The following equipment exists at present:

1. 24 frequency generators each with a frequency range from 0-15,999 Hz, stepwise in whole cycles, with seven different waveforms.
2. One noise generator with white and pink noise.
3. Four reverberation units.
4. Two 1/3 octave filter sets each with 28 channels.
5. Three ring modulators.
6. Two amplitude modulators.
7. Twenty connection panels.
8. One hundred attenuators with a signal to noise ratio of 100 dB and the strength adjustable in steps of a 1/4 dB.
9. Two digital tape recorders of the type AMPEX TIM 11.
10. Various audio tape recorders.



Digital controlled
frequency generators

A freely-chosen part of this apparatus can be set by the computer with a new value every thousandth of a second. At this highest speed about thirty parameters are able to be set at a time. If one lowers the speed to two milli-seconds, about sixty parameters can be set etc.

The computer, pieces of apparatus and the controls each have their own rooms. Composers mostly make use of the control room, but in some cases also the computer room. The control room has a console arranged in a square having an overall length of about 10 yards. Indications are made up of about 2,000 lights and the controls of a similar number of "buttons". The computer is controlled from the control room with the help of an alphanumerical and graphic terminal. This control room for composers is quiet, air-conditioned and separated from disturbing activities.

WORKING PROCESSES

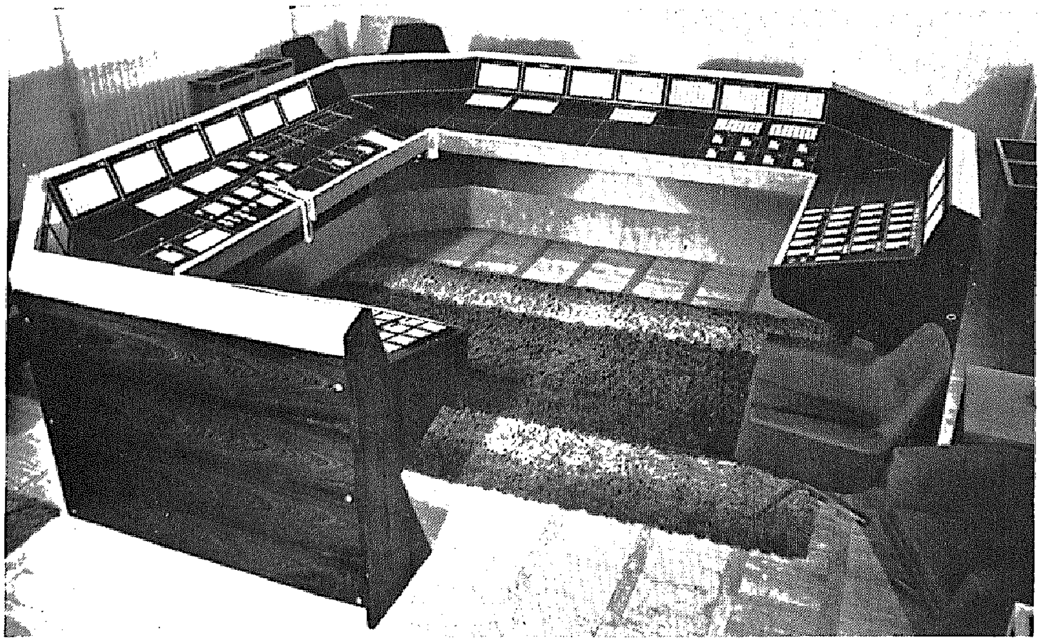
The studio has three different, but simultaneous working systems. One can work partly at the control console without the help of the computer; one can drive the computer with a foreground-background-system which means that two composers can work with the computer at the same time. The composer who is working at the control console has priority, should he desire access to the computer. The studio apparatus can thus be controlled both from the digital tape recorders, that are on-line to the computer and those that are off-line, and also directly from the computer's disk and core memories. The composer at the control board can instruct the computer from his terminal and immediately hear the result. This does not apply if the calculations are too time-consuming or if the output is too extensive as regards information. In this case, the output has to go through the digital tape recorders which entails a somewhat longer waiting-period.

THE CONTROL CONSOLE

The control console is used partly for teaching purposes, partly as an instrument for improvisation by interested composers, but first and foremost it is used for editing compositions that exist on digital tape. The composer can play chosen sections of the composition, continuously or section by section; he can re-wind quickly forwards and backwards, copy from one digital tape recorder to another and – by means of a technique that developed – add new structures while the existing are played back in real time.

The indication lights give simultaneous information on everything that is on the tape. All operations at the control console can be done without the help of the computer. This applies also to the registration of sounds and time. The feeding of the larger quantities of data that are needed when the composer wants glissandi or envelopes is done with the help of a manually controlled system.

A sequence of work at the control console, without the use of the computer, can

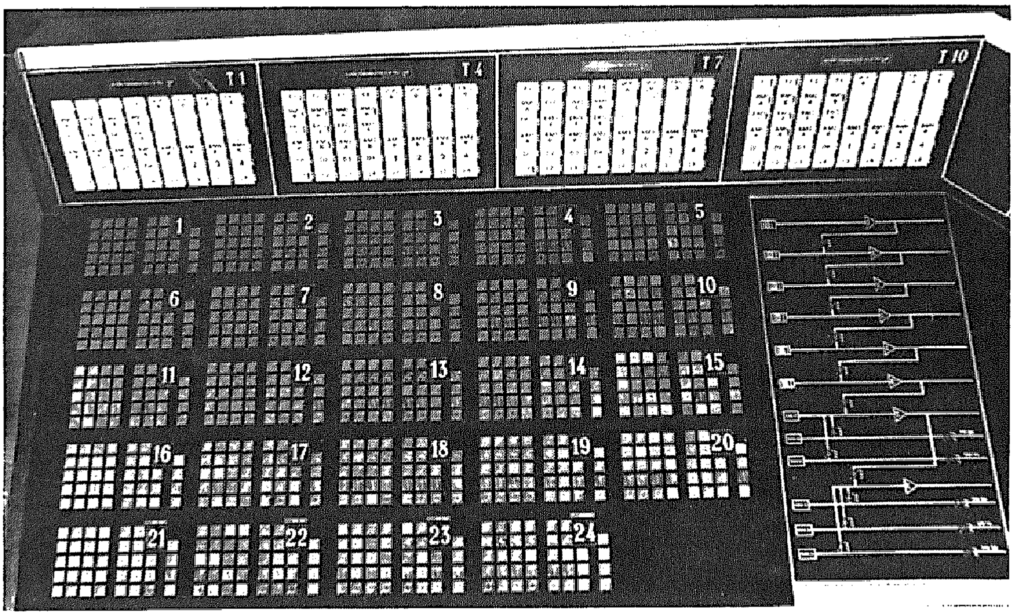


The control console

be as follows. The composer sets a frequency generator at a frequency, signal-level and wave form. The frequency generator's lights indicate the setting. The signal is coupled to an output channel. The tone's duration is set in the time register on the control console for the digital tape recorders. The tone now exists in the form of a machine setting stored in a digital register. This information is transferred to the digital tape recorder together with the time that the composer has given the tone, and also record number.

Should the composer wish to listen to the stored tone, he first gives the order "Seek the record number" and sets the number '1' in the index-register, that is the number that the recorded tape has obtained. The digital tape recorder rewinds its tape and stops at the recording number '1'. The composer gives the order "Record to buffer", after which the recorded information is transferred to a buffer-memory. The purpose of having a buffer-memory between the digital tape recorder and the apparatus registration is so as to achieve a simultaneous setting of all the studio apparatus. It takes a certain time, of course, for the digital tape recorder to play back a recording (at present about 5 milli-seconds plus 8 milli-seconds for start-stop), and if the information went directly from the tape to the apparatus registration, certain pieces of apparatus would obtain their settings earlier than others, and a certain amount of oscillation would then accompany the recorded sounds. The buffer-memory feeds all the settings in parallel the instant the clock has counted down the recorded time, and it is time for a new sound.

The composer then gives the order 'Record to console', and the information on the tone's setting is transferred from the buffer-memory to the apparatus register with the result that the tone sounds, and is indicated on the control console. If one wants to play several recordings after one another, the order "Play" is given at the



24 Digital controlled frequency generators on the control console.

same time that one gives the index register the recording number after which the tape recorder should stop. Stepwise progression of the recording occurs with the order 'Stepwise' and copying with the order 'Copy', stating the record-number with which the copying should begin and end.

A digital tape played in such a way, where each recording has a different musical structure and a given time for how long that actual setting shall last before a new structure is fed out, will in the future be able to be transferred via a computer to a digital tape with set recording times. The purpose of this conversion is that the composer will be able to make changes to the digital tape in real time, that is he can wind forward the tape to a certain point, and start playing back from that point at the same time as, for example, changing one of the sound generator's envelopes. He can also make such a change with the help of the computer. With its help, he can of course make complicated changes to the musical structure for example, all types of time or 'tempo' changes, different transpositions and types of editing.

The programming – or the notating as it was earlier called – can be divided into four types in our studio. I refer to that part of the programming that concerns the composer, not the technician. To the first type belongs the control console where the sound is defined in physical terms. The console is the improvisatory-based instrument of the composer. Completely unconcerned about the difficult problems he has been drawn into, he can quietly and without fear make his first acquaintance with electro-acoustical music here at this console. Here there are no demands, no technicians or other listeners. The console also has the role of a correction-instrument, as mentioned earlier. The synthesised sounds produce on the console a clear indication in the form of bright lights, and the composer can change without difficulty undesirable sounds in any part of the composition. Compositions which the composer has notated in the form of frequencies, amplitudes and times can also be realised in this first type of programming.

The second type of programming is also connected with the controle console, being a corresponding program. Instead of being in the studio and using the actual manual controls, the composer can sit at home and program the controls. This program is transferred to punched tape and fed into the computer, after which the composer can listen to the result. All changes that he then performs at the control console are automatically written out on the line printer so that he always has a full record of his musical experiments. With this program he can communicate with the apparatus, beginning equally well from the physical terminology as from the control console's indication system. But he can also reduce the quantity of his programming-work by making use of a system of abbreviations for certain operations and sequences, for example by calling a certain name after which one only needs to call for that name in order for the result to be coupled up to the whole series. In this way, one can store name by name until one finally calls the name of the composition, and it is played completely.

The third program-type is the possibility to use Fortran and Algol, (more need not to be said), a fourth program-level is that of simulating rules of composition technique and finally – as a fifth level – the possibility of describing sounds in psychological terms. This far, this system of description exists only in the form Pierre Schaeffer has given it in his theoretical work "Traité des objets musicaux". We at EMS shall try if given economical possibility to realize the idea in terms of a computer program. Here is a basic outline of the way we intend to deal with this problem.

Our starting-point consists of selected sound objects which are recorded on an analog tape. These sound objects are given a digital form by means of an analog to digital converter. In this form the computer can give an analysis of the sound in physical terms. Composers and researchers can now take away those parts of the sounds to which the ear does not react and in this way find the least possible amount of information which is needed in order to synthesize a similar sound object. A test panel can now begin to compare the original and the synthesized sound. The test panel gives its opinion about the sound in the psychological terminology invented by Schaeffer, and, with access to both the sound's physical data and psychological data in conjunction with a good experimental situation, where one can change in actual time the sound's different parameters, we will try to bridge the gap between the physical and psychological descriptions of a sound object. Should we succeed in giving a sound object such a description, then the next step is to try to build "scales" between two such sound objects by allowing the computer to change the physical properties of the sounds. A test panel will search for corresponding changes in our experiences. In this way we hope to be able to construct a psychological system of description, in which the composer writes the desired sound within the framework of a number of psychological variables. The composer no longer plays with a keyboard, and he no longer presses buttons. He writes his sounds and musical structures in psychological terms, and the apparatus at EMS translates these terms into sounds.

But while we are waiting for this, we are compelled to use a programming-language which is apparatus-orientated. Our version of this (that is, programming type 2) is called EMS-1. This language corresponds to the control console; that is, it describes the setting of the apparatus and data that can be performed at the console.

The input text such as frequencies, amplitudes, connections between various devices and so on, may be present on any input media except magnetic tape. Thus a user may punch his text on an off-line teletype and the resulting papertape can be read in by the papertape reader on the computer. This text is translated into the internal digital form used by the studio (compilation), i.e.

The EMS-1 programs check the input text for errors and produces output in the form of studiorecords on magnetic tape and output text with the lines numbered.

It is possible to run the EMS-1 programs interactively i.e. it is possible to produce a record on the magnetic tape, listen to the result and if no good make corrections and additions to the input text, listen to the result once more and if good save the record listened to and continue to make new ones. The length of a piece is unlimited, the user just adds new records to the previous ones.

The language contains conditional statements which make it possible to conditionally control the compilation of the input text. The "MACRO" option makes it possible for the user to reduce the amount of input text.

The setup of the studio is done nearly in the same way in the EMS-1 language as should be done in the studio control room with the control panel. The user specifies the sound producing sources, the sound modifying devices and the connections between these and the output channels. To this the user can add enveloppe curves and glissando curves to all frequency generators. The input text is translated to so called "SORTRECORDS" and stored on the 'TEMP' disk (temporary disk unit). It's possible to listen to the sounds generated from these sortrecords with a 'PLAY' command. The sounds on the 'TEMP' disk is called a sound object. When the user is satisfied with a sound object he can transfer it to another disk unit called the 'MIX' disk which contains the sortrecords from one or more sound objects. These can eventually be recorded on magnetic tape (MT) together with a MT-'LABEL', which makes it possible to find the block later on. Within a block a local time is specified, which starts with zero at the mix disk with a local time specified i.e. the sound object is to start at this time relative the start of the mix block start. A sound object can be transferred more than once to the mix disk with different local times each time. The EMS-1 text must not be written in time sequence.

The use of the EMS-1 system can be arranged differently depending on whether it is interactive or non-interactive, and whether it is an initial or corrective run.

An initial run is performed the first time the input text is presented to the computer. The program will produce both a magnetic tape with EMS code, used to play the piece on the studio, and an output text, which is the recorded input text with line numbers inserted.

During a later corrective run the output text from the previous run may be used as input and corrections added in an interactive mode. Alternatively, the EDIT utility program can be used to insert alterations before the text is recompiled. The line numbers are used as reference points for find commands in the EDIT program.

During an interactive run, text is typed in at the display terminal (TV) or possibly a teletype, and messages about any formal errors are returned to the terminal which is then placed in corrective mode. The use is expected to correct the error immediately. It is also possible to have the last setup, or any sequence of setups from the piece, to be played back.

In non-interactive mode, an entire sequence of text has been prepared beforehand, either by punching a paper tape off-line or by use of the PIP and EDIT utility programs. The text is then compiled at the computer from beginning to end, and any error messages will be listed on the line printer or a teletype. This has the advantage that the EMS studio need not be available and that the computer time used is considerably less. The errors can then be corrected by use of the EDIT program or during an interactive, corrective run.

A normal way to work interactively could be to specify a sound object. Listen to the result. If not satisfactory, change some of the parameters. Listen to it again and so on until the sound object is satisfactory. The object can now be added to the previous objects on the mix disk with a "MIX" command. It is possible to listen to the mix block with a "PLAY(MIX)" command. It is possible to overwrite sortrecords on the mix disk with sortrecords from the 'TEMP'disk. The rule is: if two sortrecords are given to the same device at the same local time, the sortrecord inputted last will be kept and the other(s) taken away. When a mix disk block is satisfactory it can be recorded on the MT with an 'END' command. Note that the commands PLAY, PLAY(MIX) also generate codes on the MT but code to be permanently recorded is recorded with the 'END' command and thus always from the mix disk.

The EMS-1 program is developed in cooperation with UDAC (Computer center of Uppsala University, Dr. Klaus Appel, Box 2103, 75020 Uppsala). The following example is made by Torbjörn Höglund, same address.

```
BEGIN(STORS)
GØ=196;AØ=22Ø;DØ=147;HØØ=123
C1=262;D1=294;E1=33Ø;F1=349;HØ=247;AØ=22Ø;CØ=131
FØ=175;GØ=196
NOT1=1ØØØ;NOT2=5ØØ;MAX=8Ø;WF=2;ET1=-1;ET2=-2
TON="LT(GT1,GT2)MAX1=MAX+1Ø;T1=TID/1Ø;T2=TID-2*T1;T3=2*T1
GT2=GT2+TID;GT3=GT2-1ØØØ;IFPOS(GT3)GT2=GT3;GT1=GT1+1;<
FR1=FR+1Ø
IFDEF(LISTA)WRITE(MAX1,T1,T2,T3,GT,FR1)MESS(???);<
FG(NR,,,WF)>ENV(5Ø,MAX1,T1,2)>ENV(MAX1,MAX,T1,ET2)>T(T2)
GLIS(FR1,FR,T3)
FR1=FR*2;NR1=NR+1;MAX1=MAX-6
FG(NR1,FR1,,WF)>ENV(5Ø,MAX,T1)>ENV(MAX,4Ø,T2)
FR1=FR1*2;NR1=NR1+1;MAX2=MAX-1Ø
FG(NR1,FR1,,WF)>ENV(5Ø,MAX1,T1,ET1)>ENV(MAX1,MAX2,T1)>ENV(MAX2,3Ø,T2)"
SILL="TID=NOT1;FR=C1;TON
TID=NOT2;FR=D1;TON
FR=C1;TON
TID=NOT1;FR=HØ;TON
FR=C1;TON
FR=E1;TON
TID=NOT2;FR=F1;TON
FR=E1;TON
TID=NOT1;FR=D1;TON
FR=E1;TON
FR=C1;TON
TID=NOT2;FR=AØ;TON
FR=FØ;TON
TID=NOT1;FR=GØ;TON
FR=CØ;TON
"
FG3>FG6>CHA(1,1ØØ)
FG9>FG12>CHA(2,1ØØ)
FG15>FG18>CHA(3,1ØØ)
FG(21)>FG21>CHA(4,1ØØ)
FG(2Ø)>FG21
FG(24)>FG24>CHA(4,1ØØ)
FG(23)>FG24
FG(22)>FG24
GT1=1;GT2=Ø
NR=1;SILL;SILL
NR=7;GT1=5;GT2=Ø;SILL;SILL
NR=13;GT1=9;GT2=Ø;SILL;SILL
NR=19;GT1=13;GT2=Ø;SILL;SILL
MIX
END
```

DESCRIPTION OF THE EMS1-TEXT.

'SILL' IS A MACRO FOR A 15 NOTE CANON. FOUR DIFFERENT SOUNDGENERATOR GROUPS EACH PLAY THIS TUNE. THEAY START TO SOUND AFTER RESP. 1,5,9,13 SECONDS. ONE NOTE IS BUILT BY THE MACRO 'TON'. THIS MACRO CONTAINS A GROUND NOTE FOR SOUNDGENERATOR NR "NR" AND 3 OVERTONES ON SOUNDGENERATORS "NR+1", "NR+2" AND

"NR+3".
EACH OF THE TONES HAS SPECIFIC ENVELOPES TO GIVE THE WHOLE TONE A SPECIFIC CHARACTERISTIC. THE 'TON' MACRO STARTS WITH A CALCULATION OF DURATION TIMES OF THE ENVELOPES. IF THE SYMBOL 'LISTA' IS DEFINED THE CALCULATED VALUES OF THE SYMBOLS MAX1, T1, T2, T3, GT1 AND FR1 ARE WRITTEN OUT AND AFTER THAT THE MESSAGE "???" IS WRITTEN TO THE USER AND THE SYSTEM WAITS IN ERROR MODE. THIS GIVES THE USER AN OPPORTUNITY TO CHANGE VALUES. BY A "DELETE(LISTA)" THIS MESSAGE WON'T BE WRITTEN OUT UNTIL 'LISTA' IS REDEFINED. THE CONNECTIONS ARE MADE SO THAT SOUNDGEN. 1-6 ARE CONNECTED TO CHANNEL 1, 7-12 TO CHANNEL 2, 13-18 TO CHANNEL 3 AND 19-24 TO CHANNEL 4. THE 'SILL' MACRO "PLAYS" THE CANON ONCE THUS EACH GROUP "PLAYS" THE TUNE TWICE. THE 'MIX' COMMAND TRANSFERS THE 'TEMP' DISK TO THE 'MIX' DISK. THE 'END' COMMAND IMPLIES GENERATION OF MT-CODE ON THE MT. THE BLOCK LABEL ON THE MT WILL BE 'STORS'. THIS LABEL CAN BE USED IN A LATER 'PLAY' COMMAND.

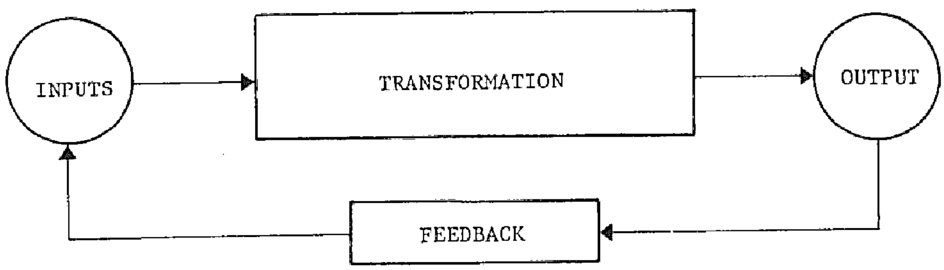
COMMENTS LINE BY LINE

LINE	COMMENT
1	BEGIN A NEW BLOCK WITH LABEL 'STORS' ON MT AND NAME 'STORS SRC' ON THE DT FILE ON UNIT 5.
2-4	ASSIGN FREQUENCY VALUES TO THE SYMBOLS G0,A0 ETC.
5	NOT1, NOT2 ARE TIME VALUES OF A WHOLE NOTE RESP. A HALF NOTE. MAX IS USED IN ENV TERMS ON LINES 10,13,15. WF IS WAVEFORM FOR ALL THE 'FG':S ET1,ET2 ARE CURVEFORMS OF THE ENV TERMS ON LINES 10,15
6	START OF DEFINITION OF THE 'TON' MACRO. THE TONE STARTS AT GT1 SEC. GT2 MS LOCAL TIME. MAX1 IS MAXIMUM AMPLITUDE OF THE FIRST FG. T1=1/10, T2=8/10, T3=2/10 OF THE WHOLE TONE TIME 'TID'.
7	ADD THE TONE TIME 'TID' TO GT2. IF GT2 MORE THAN 1000MS (1 SEC.) THEN GT3 WILL BE POSITIVE AND 1 IS ADDED TO GT1 (1 SEC.) AND 1000 MS (1 SEC.), I.E. THE SAME AMOUNT IS SUBTRACTED FROM GT2.
8	FR1 IS THE FREQUENCY USED IN THE GLIS TERM AT LINE NR 11.
9	IF THE VARIABLE 'LISTA' IS DEFINED THE VALUES OF MAX1,T1,T2,T3,GT1, FR1 IS WRITTEN OUT. AFTER THAT THE MESSAGE '???' IS WRITTEN OUT AND THE PROGRAM WAITS IN ERROR MODE FOR INPUT FROM UNIT 4 (TT OR TV). A 'DELETE(LISTA)' WILL MAKE THE VARIABLE LISTA UNDEFINED AND THUS MAKE THAT NO MORE OF THE ABOVE PRINTOUTS WILL COME OUT.
10	THE FIRST FG GOES FROM 50 TO MAX1 DB IN T1 MS CURVEFORM 2, FROM MAX1 TO MAX IN T1 MS, CURVEFORM ET2 AND STAYS AT MAX DB FOR T2 MS.
11	FIRST FG GOES FROM FR1 TO FR HZ IN T2 MS.
12	FR1 IS THE FREQUENCY OF THE FIRST OVERTONE. NR1 IS THE NUMBER OF THE SECOND FG. MAX1 IS USED IN THE ENV TERM ON LINE 15.
13	THE SECOND FG GOES FROM 50 TO MAX DB IN T1 MS, CURVEFORM 0 AND FROM MAX TO 40 DB IN T2 MS.
14	FR1 IS THE FEQUENCY OF THE THIRD OVERTONE USED IN THE 'TON' MACRO. NR1 IS NUMBER OF THE THIRD FG. MAX2 IS USED IN THE ENV TERM ON LINE 15.
15	THE THIRD FG GOES FROM 50 TO MAX1 DB IN T1 MS, CURVEFORM ET1, FROM MAX1 TO MAX2 DB IN T1 MS AND FROM MAX2 TO 30 DB IN T2 MS.
16	START OF THE DEFINITION OF THE 'SILL' MACRO WHICH CONTAINS THE WHOLE CANON "STORSILL OCH SMA SILL".
16-30	TIME AND FREQUENCY OF THE TONES IN THE CANON ARE GIVEN AT EACH LINE. NOTE THAT THE LOCAL TIME IS ADDED WITH THE TIME VALUE OF THE NOTE EACH TIME THE MACRO 'TON' IS CALLED SO THAT THE TONES WILL COME IN THE RIGHT TIME SEQUENCE.
31	END OF THE 'SILL' MACRO.
32-39	FG:S 1-6 ARE CONNECTED TO CHANNEL 1,FG:S 7-12 TO CHANNEL 2, FG:S 13-18 TO CHANNEL 3 AND FG:S 19-24 TO CHANNEL 4.
40	STARTTIME 1 SEC. AND 0 MS FOR THE FIRST 'SILL' MACRO.
41	FG:S 1-3 PLAYS THE CANON TWICE.
42	FG:S 7-9 PLAYS THE CANON TWICE STARTING AT LOCAL TIME 5 SEC. 0 MS.
43	FG:S 13-15 PLAYS THE CANON TWICE STARTING AT LOCAL TIME 9 SEC. 0 MS.
44	FG:S 19-21 PLAYS THE CANON TWICE STARTING AT LOCAL TIME 13 SEC. 0 MS.
45	TRANSFER THE SORTRECORDS PRODUCED BY THE ABOVE 'SILL' MACROS ON THE 'TEMP' DISK TO THE 'MIX' DISK. ("TRANSFER" HERE ALSO INCLUDES SORT)
46	MAKE MT CODE FROM THE MIX DISK. ERASE THE MIX AND TEMP DISKS UNLESS ANY OF THE COMMANDS KEEP, KEPP(MIX), KEEP(ALL) COMES AFTER THE END COMMAND. THE PROGRAM NOW EXPECTS A NEW BEGIN COMMAND TO START A . NEW BLOCK.

.EJECT

An example of the fourth level of computer programs which I mentioned earlier, is "Musicbox" developed in cooperation with the programmers David Fahrland, USA, and Kaj Beskow at EMS.

The philosophy of the Musicbox is to recognize everywhere the systems concept of



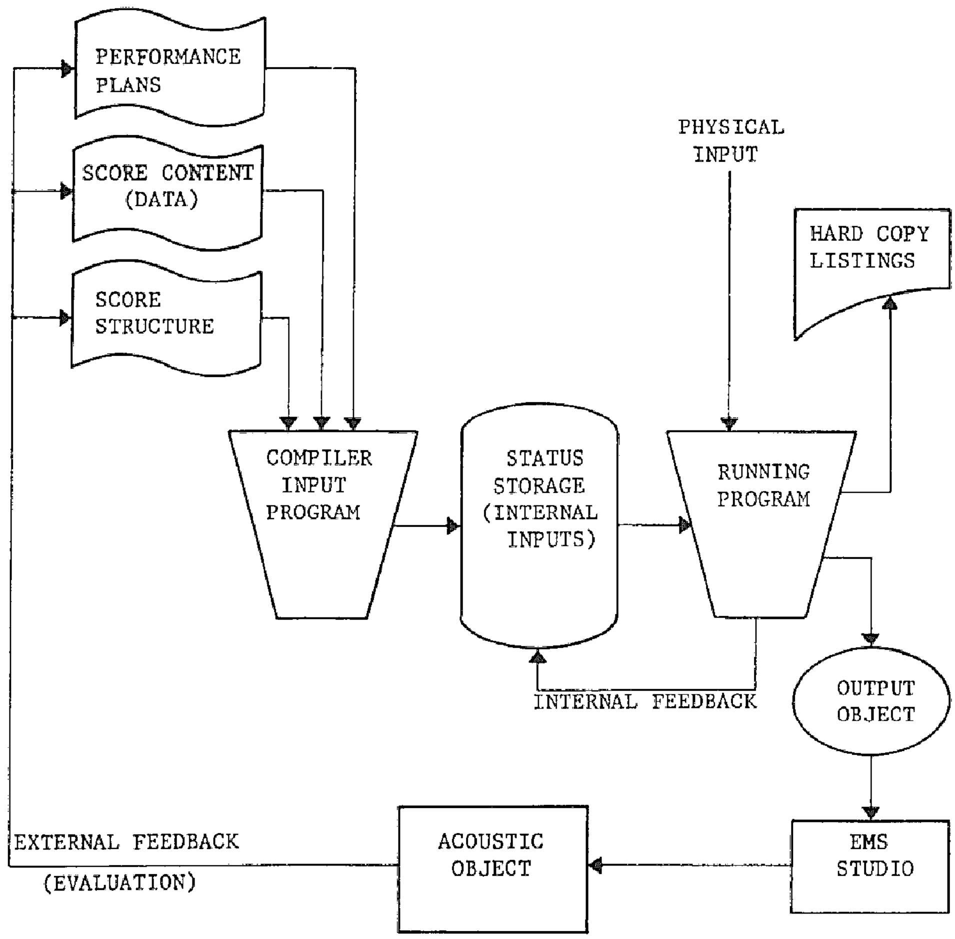
This idea is applied uniformly from the highest level of conceptual hierarchy to the lowest level of implementation. This methodology of problem representation has been well developed in the Operations Research and Engineering Systems Sciences over the past twentyfive years, and now has come into being in the areas of Socio-economic Sciences and hence must be reckoned with as one of the most powerful present day forces.

The INPUTS to Musicbox may consist in general of any SET of elements. Alphanumeric symbols which may specify Score Structure, Score Content, or Performance Plans are inputted to the Compiler part of the program and translated into an internal memory representation. These symbols may represent such diverse things as studio parameter values, studio patching, pre-defined "box"transformations, box connections, and many other things. The transformational or "running" part of the program then repetitively performs computations on internal memory to produce a new status of memory by feedback and an OUTPUT object. The OUTPUT object is in the form of a fixed format tape of parameters which can be acoustically realized on the EMS studio. An auxiliary output object in the form of a printed listing is available for visual verification of the result.

BASIC ELEMENTS

The acoustic waveform may be described and manipulated in the terms of its parameters frequency, intensity, and time duration. The magnitude of these parameters is represented by some real *value*. The continuous changes of these *values* with *time* describe *envelopes* of a parameter. The discrete changes of these values at distinct time instants describe *events* of a parameter.

The Musicbox stores values as box *inputs*, box *parameters*, and *box states*. Each change of value, whether discrete or continous (sampled), is represented by a *message* from some box *output* to another box input. A message is conceived of as a value plus event as the philosophy of the musicbox is that there can be no change



without some associated value or conversely no value may exist unless it can change.

Boxes which transform single messages (or the "flow" of a message chain) are defined by selection from a pre-determined menu (of micro *box types*). Structural *connections* of these boxes and selection of fixed values (data) determine the function of the composition system.

Groups of boxes which perform a standard function, may be combined into a new *structural unit* known as a *macro* box. *Macro* boxes can also "feed" information to passive parameters of other boxes, which then defines a structural *hierarchy*.

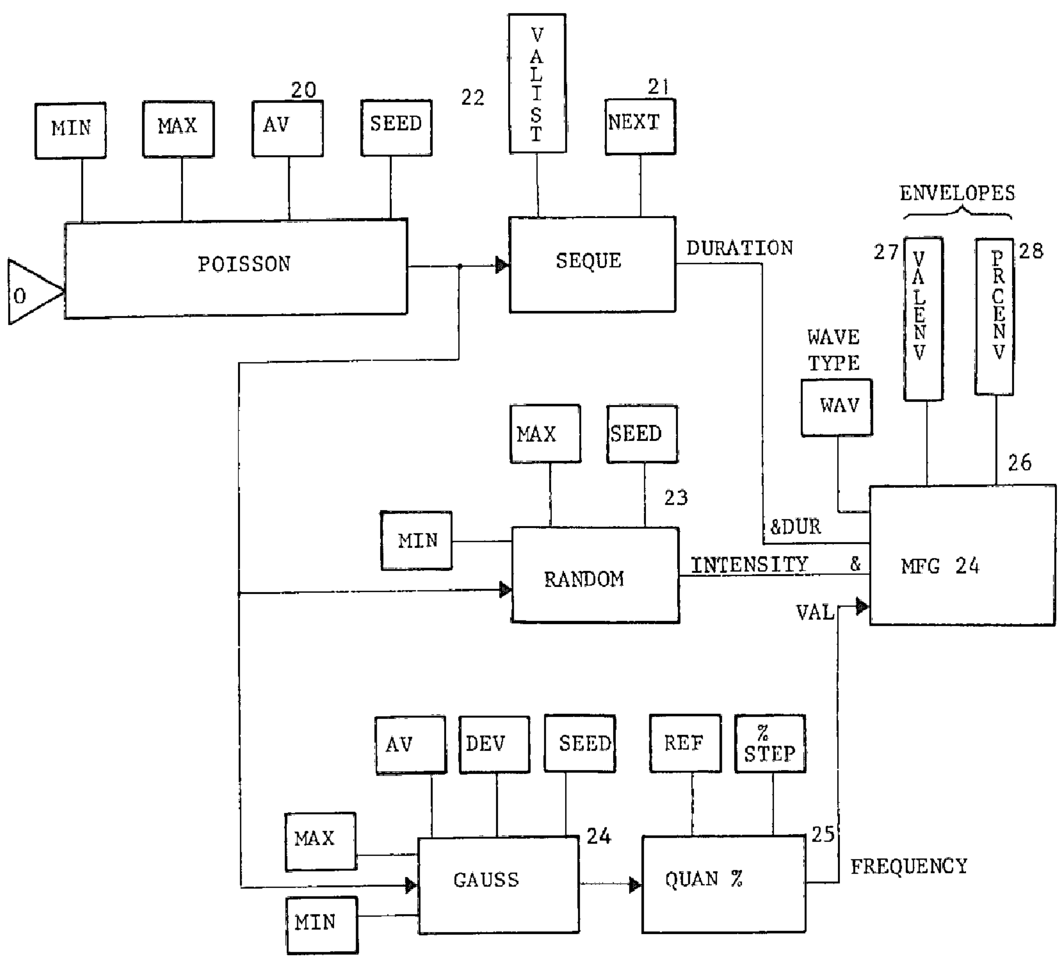
Finally, *command* information is given which plans how the above system is to function in time and thus produce an intermediate result of messages and a final acoustical result of studio tapes.

A working process with Music Box could be as follows:

00. The "atoms of the system" are events with attributes of time and space called MESSAGES.

0. The "building stones" of the system are miniature systems which input, transform, and output messages called micro BOXES.

1. The composer makes a consolidation of interconnected micro BOXES into sub-assemblies called MACRO boxes.
2. Interconnected MACRO and micro BOXES are grouped and represent some musical or acoustical concept called a STRUCTURE.
3. DATA values are attached to structural parameters in order to give some definite shape and meaning to the STRUCTURE.
4. The composer prepares COMMANDS for instruction of the runtime program to coordinate the implementation of "drawings" according to a time schedule.
5. Grouping of many sub-plans together into an overall PLAN.
6. Simulating the scaled structure according to plans to produce intermediate results (internal messages) and the END PRODUCT (records to the studio).
7. Collecting final statistics from the implementation process and delivering a completed product (tape) and associated listings.
8. Criticism of the audio product followed by feedback to any stage of the construction process and repetition of this procedure.



MUSICBOX EXAMPLE

The Musicbox example shown above is merely constructed to illustrate some of the boxes that are useful in an aleatoric composition technique. It generates a number of tones with pseudorandomly chosen intervals, frequencies and intensities and the durations are obtained from a cyclic value sequence.

BOX DESCRIPTION

The numbers below refer to the box numbers in the figure. SEED parameter in some of the boxes below is the startingvalue for the pseudorandom sequences used.

20. POISSON

After being activated once at the input the microbox POISSON generates events which have time intervals in the range MIN to MAX milliseconds, exponentially distributed with the average interval AV. The number of messages delivered per time unit from the POISSON box thus is approximately Poisson-distributed.

21. SEQUE

When activated at the input, the microbox SEQUE transmits a duration value from a cyclic duration sequence stored in the list VALIST (22), starting at the list index NEXT.

23. RANDOM

When activated at the input, the microbox RANDOM delivers a pseudorandom intensity value, uniformly distributed in the range MIN to MAX decibels.

24. GAUSS

The microbox GAUSS delivers when activated at the input a pseudorandom frequency in the range MIN to MAX Hertz, approximately normally distributed with the average AV and the standard deviation DEV.

25. QUAN %

When activated with a continuous Hertz value at the input, the microbox QUAN% outputs a logarithmically quantized Hertz value. The scale reference frequency is REF and the octave scale step is $^2\log(1+ \%\,\mathrm{STEP})$.

26. MFG24

The macrobox MFG24 distributes tones with specified frequency VAL, intensity INT (100% reference for envelope), waveform type WAV, duration DUR and envelope to the 24 frequency generators of the studio. The envelopes are specified by two lists, namely the intensity percentage list VALENV (27) and the duration percentage list PRCENV (28). These lists can be changed from tone to tone. In the example above the envelope lists and the waveform type are common to all tones.

INTRODUCTION TO SOUND PROCESSES DEVELOPED AT EMS

During approximately the same period as EMS built its studio, the electronics industry was revolutionised by the fall in price of integrated circuits – especially the digital – which sank lower and lower in cost at the same time as they became all the more complex.

Since EMS also had as its task the development of new apparatus, work began in the spring of 1970 to develop a system for treatment by using the highly integrated digital technique would combine a fully programmable mixer with synthesizing possibilities. In certain respects, its similarity to a modern computer is remarkable. It can collect up material and supply its own, treat it and deliver it. We have provisionally called the apparatus a "sound-processor". The technical designer, Björn Sandlund, gives the following description of the processor. Every sound in the sound-processor is represented by an electric voltage of between -10V and +10V, This sound-signal is permitted to assume all the possible values between those limits. The sound-signal's variation in time will be analogous to the variation in airpressure, i.e. a direct analogy to the sound waves.

1.1. Digital Signals

All control of the sound-processor's functions can be done with digital signals. The digital signals in the sound-processor can only have two values, OV or +5V. This means that the control becomes "quantatized", i.e. none of the control functions are continually variable, but can only have certain unobtrusive values. Thus for example the level for a single rule can only be set at 256 different values, despite the fact that the potentiometer track itself is stepless. A sound generator's frequency can have 16.384 values etc. The connection between the digital control signals and the analog components consists of an analog switch. Its function can be linked to that of a relay, but it is much quicker, has low power consumption and is small in dimensions. Two digital cassette tape recorders are used for storing the digital data. One cassette tape can store control data for the sound-processor for more than half an hour's complicated electronic sound structures.

1.2. The cassette box, the system's smallest component

All the electronic components in the system are housed in a standardized box-container of a cassette type. The frontal measurements concur with those traditional for mixer-consoles, 40x190 mm. Other measurements and the mechanical workmanship are however wholly adapted to the special needs of the system.

We have tried to give each cassette box a well-defined function that is carried out just by those components that exist in the box. This type of purely technical manufacture has the disadvantage that some boxes tend to be crammed full with electronics, while others are less well utilised. However, we gain in increased understanding and lucidity at the same time that routines for tracing faults are made easier.

The boxes are so arranged that they can be easily connected to or from the system so that one can quickly adapt the sound-processor for a certain application.

1.3 The cassette box's system

Page 158 shows a picture of how the cassette boxes fit into the sound-processor's frame. The boxes are arranged in rows with twenty-eight to each row. The dark panels show the places where the "sound-carrying" boxes are placed. These are called SOUND BOXES. The largest area comprises of forty-eight sound boxes, devided into sixteen "channels", each with three sound boxes. The smaller area consists of twelve sound boxes divided into four channels, with three sound boxes to each channel. This panel is principally reserved for the main outputs from the sound-processor, but is in many respects fully comparable to the larger panel. The three remaining panels are reserved for fixed electronic controls, indications of signal-levels etc. These cassette boxes are called DATA BOXES.

Let us examine one of the twenty channels. In the uppermost of the three compartments is placed a sound box of a sound source character (e.g. a frequency generator, a noise generator, or a pulse generator). One can also permit the actual sound source to come from outside the sound-processor (e.g. from a microphone, a tape recorder, or an external generator). One must then set a suitable input amplifier in the uppermost compartment. One can in this way connect an external sound source and adjust the signal-strength by means of the input amplifier. For further on this, see the section concerning the sound box.

The signal is coupled from the sound source to the middle compartment. This is specially reserved for sound boxes that treat or re-work the sound (e.g. filter, echo, chopper, compressor or expander).

The signal is then coupled on further to the lowest compartment, which is set aside for sound boxes regulating the signal strength, e.g. attentuators or special envelope generators.

The signal then goes out from this compartment and out of the processor via a cannon-connection on the back of the frame. This signal is sufficiently powerful to go for a long distance (stipulated signal swing over 100 ohm. Figure 1.3.1 shows how a channel is built up.

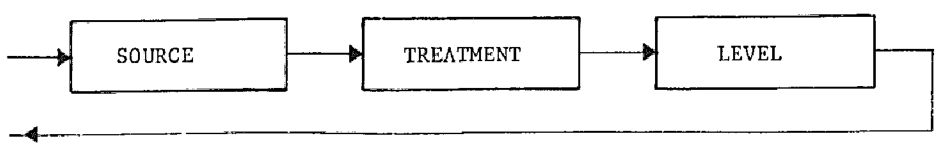


Fig. 1.3.1

We should be able to accomplish a number of the simplest assignments on the processor with the connections described above. We should be able to connect twenty microphones, treat the signals from these, add a signal level and record the result on a twenty channel tape recorder, if we had one. We should also be able to put in twenty frequency generators, give them a signal level and run twenty loudspeakers. By programming the controls of the frequencies and amplitudes, we should be able to use the connection for the playing back of advanced electronic sound structures.

1.4 The concept of simple sound sources, and compound sound sources

We should, however, soon reach a situation in our work with the sound-processor, where we would want to reduce the number of channel outputs by joining them together to form new compound signals. By allowing the signal from a channel output to branch out from a contact at the bottom of the sound-source row, the first sixteen channel outputs are made available for all the sound source compartments. We can now use which signals we want in just one sound source compartment if we add a so-called mixing amplifier. This is a sound box which is provided with sixteen switches. These are connected with the sixteen output channels. (See the picture.) If, for example, we connect a mixing amplifier to channel 10, sound source compartment, and connect the switches for channels 1, 5 and 15, then we have a sound source on channel 10 that is a compound of channels 1, 5 and 15.

Channels 17-20 will certainly for the most part be used as output channels, and therefore house mixing amplifiers in their sound source compartments. Figure 1.4.1 shows how each channel output branches out from the sound source compartment.

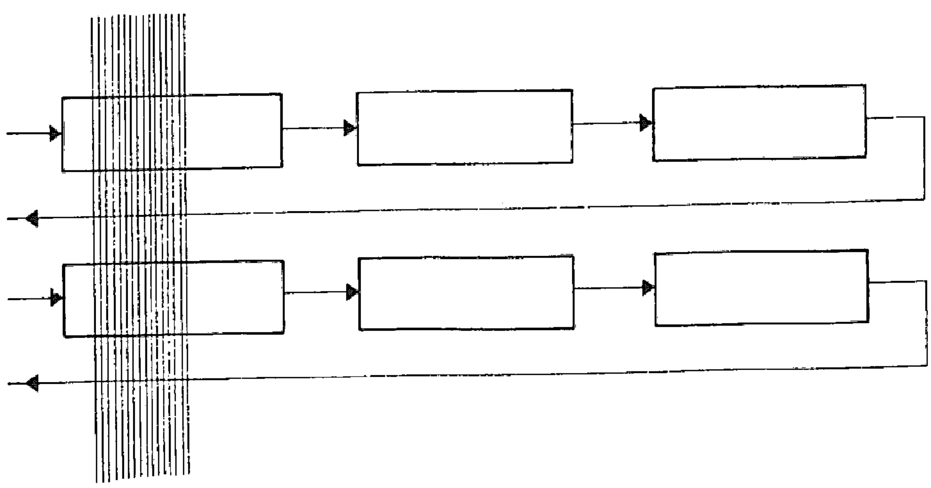


Fig. 1.4.1

1.5 Complete audio paths

We shall now complete the picture of the paths, page 159. Here one sees that the treatment compartment also has available output channels. As a result, it is possible to use the treatment apparatus in an analog-controlled way. For example, it can be used as an amplitude modulator, which is controlled by signals from another output channel. Furthermore, we see that the treatment compartment has a special "tapping-bus" consisting of eight outputs and eight inputs. This bus can be used for studios that have already invested in external treatment apparatus, e.g. echo, octave filter. The signal can thus leave the processors' sound-source, pass out to the external treatment equipment, and return to the processor's signal strength regulating compartment. The connection is accomplished by putting in a special sound box in the treatment compartment. It is possible to connect eight external treatment devices to the processor at the same time.

Finally, we see on page 159, that the sound source row has its own input bus of sixteen connections, connected to multi-way connector on the back of the processor. This bus can be used to connect up in-coming signals from tape recorders. Thus, for example, one can have a sixteen channel tape recorder permanently connected to the processor via this contact. The channels are connected to sound sources in the processor by adding special mixing amplifiers in the sound source compartments.

It is also evident from the picture on page 159, that the output signal from channels 1-16 are accessible on a multi-way connector.

1.6 Listening monitor and level indication

We shall now go into the function of the fixed equipment of the processor.

Below each channel-row of the processor, we can see a round press button (page 158). By pressing any of these, we can connect the signal from the corresponding channel to a special monitor amplifier. The working of the button can be electronically connected between momentary operative-connections as long as the button is kept pressed down, and alternative-connections are established and broken every other time one presses.

There are four monitor amplifiers, placed on the left side of the channels (see page 160). With these, one has the possibility of varying the signal strength for the monitoring of channels 1-16 with one control, and channels 17-20, with separate controls. The signals that have been connected via the round press-buttons are also connected to each monitor amplifier.

Normally the output is connected from each monitor amplifier to its own monitor loudspeaker. But one can also be connected to the monitor amplifier's outputs directly by means of head phones, connected to the front of the monitor amplifiers. In this way, one can have four pairs of head phones with separately prepared signal levels on the connected channels at the same time. See the picture of monitor amplifier on page 160.

The signal strength is indicated on a TV-screen in the form of twenty columns, whose height is a measurement of the top value for the signal. To exceed the signal's permitted dynamic range renders the TV-picture negative, and one obtains a clear indication that one has over-loaded the equipment.

Because the over-loading level is lower on many tape recorders than on the processor itself, the level, at which the over-loading indication begins, has to be made variable. Furthermore one must have the possibility to make the indication of overloading dependent on the frequency so as to prevent one from over-loading the tape-recorder at high frequencies. (The tape recorders that are used can, for example, be CCIR or NAB corrected).

If one has any of the round monitoring buttons connected, the corresponding columns on the TV-screen becomes ribbed. In this way, it becomes easier to look at the right column.

1.7 Manual operation

It has been pointed out earlier that the processor also has the conventional qualities of a mixer. This is best clarified by describing how the processor is operated in a purely manual way.

On page 163, there is a picture of the processor's control unit CONTROL BOX 1009. On page 158, we see its position in the processor (D6T). We have put on the power supply, and pressed PLAY on the control unit; now all the sound boxes are manually operable. The sound continues until we press STOP. If we desire a certain duration for a particular sound, we can throw the flip-flop coupler Duration from the position " ∞ " to "FROM CLOCK". We can now put the duration on the keyboard on the processor's clock, see page 163 and its position on page 158 (D6L). The sound now lasts only for the time indicated on the clock's number indicator.

1.8 Recording of the box settings on digital tape cassettes

There are two digital cassette tape recorders in the system. The exterior reminds one a great deal of a conventional audio-cassette tape recorder, but it has a somewhat different tape drive system in order to render a short start and stop time possible. Naturally, this means that electronically speaking, it is entirely different from an audio-cassette tape recorder. The digital cassette tape recorder is shown on page 161. The cassette tape recorders are operated from a special data box, TAPE COMMANDS 1015 (see page 163). This apparatus is dominated by three round press-buttons; READ, WRITE and COPY.

Recording on the cassette tape occurs as soon as we have pressed WRITE and the processor sounds (i.e. PLAY on CONTROL BOX 1009 is engaged). Even at this point, we can distinguish between two recording operations. With the duration switch in position ∞ on the CONTROL BOX, page 163, the recording will proceed continously until we press STOP. It is also possible to record continous changes from the sound-box settings at the same time as listening to these in real time.

With the duration switch in the position FROM CLOCK, we can build up a sound-box setting step by step and give each step a certain time which we set on the processor's clock. When the digital data is played back via the sound-boxes, the settings appear as an even stream without those pauses made at the recording when the settings were changed. This step-wise method also gives the possibility to record very quick and complex settings, which would be impossible to realise in a real time recording method.

If, for some reason, we are not satisfied with the real time scale which the recording has, we can easily "adjust" the recording's electronic clock, see page 163, so that the duration for the apparatus setting on play-back becomes different from that at the recording. The speed can be varied ten steps either side (slower or faster) of the normal. The time scale is shown by number indicators on the recording's electronic clock, TIME SCALE BOX 1017.

1.9 Control of the sound-boxes from the digital cassette-tape recorder

When we have recorded our apparatus settings on cassette tape, we have the possibility to allow this data control the settings on our sound-boxes. However, the sound-boxes' knobs and buttons cannot be moved, which in itself could have been effective.

Control from the cassette-tape begins when we press PLAY on CONTROL BOX 1009, and have pressed the button READ, activated by TAPE COMMANDS 1015.

The feeding-in of data into the processor can take place in an even flow if we so desire. We can also cause the cassette tape recorder to break off in various ways with the help of the HOLD MODE knob on CONTROL BOX 1009. In the position INSTR, further feeding-out of data from the cassette tape recorder is cancelled until TIMER 1015 has obtained information concerning the duration of the sound-box's setting. It is necessary to press PLAY again in order to obtain the next group of settings. In this way, it is possible to make a detailed study of a recording, or make use of an operation which manually controls the speed of the feeding-out of the new sound-box settings.

We shall return to the other positions of HOLD MODE in the section entitled "Programmed Editing".

1.10 Copying of digital data

This operation requires the loading of tapes onto both cassette tape recorders, and that one of the tape cassettes has the required data already recorded onto it. The lights of TAPE COMMANDS 1015 indicate whether we transfer data from tape recorder A or B. (The cassette recorders are labelled A or B.)

The operation of copying is performed when we press COPY on TAPE COMMANDS 1015, and then press PLAY. In reality, this operation is a combination of READ and WRITE operations.

If we wish to copy data from cassette tape A to B, the data from A is first deciphered in the soundboxes, and then immediately recorded from there onto cassette tape B. The reason for this rather complicated copying procedure is explained in the next section.

1.11 Modification and Additions during the copying of digital data

It is a basic requirement to be able to make alterations and additions to a digital recording smoothly. Audio-techniques have made use of many channeled tape recorders, where one can record one or several channels at a time, add new channels, or mix together several channels into one, etc. However, one is tied to working in real time with this method. In a similar method with addition and modification of the digital recording, we have the possibility of accomplishing these stepwise.

Suppose we have done a recording of an apparatus setting in order to produce some electronic sound structures. However, we are dissatisfied with the amplitude on e.g. channel 1. The data is "deciphered" from e.g. cassettetape A to the sound boxes and its content is recorded onto cassette-tape B.

We now want to prevent channel 1's signal regulator from receiving data from cassette-tape A. Instead, we want the regulator to receive new data from the operator. (He moves the slider control). This is brought about when one connects the regulator from channel 1 to LOCAL-MODE, which takes place when one lifts up the sound-box's upper handle. See the figure on page 156.

It is possible to have an arbitrary number of handles lifted up in the LOCAL position, and also make whatever changes and additions during copying as many times as one likes.

1.12 Programmed Editing

In the previous section, we have seen how one can do modifications and additions to the digital recording. When composing electronic music structures, one very often wishes to manipulate one's material in a more detailed fashion. For example, one might want to change the position of two structures, erase part of the composition, duplicate certain structures and put them in between others. Perhaps one might also want to combine purely "synthesized" structures with other "concrete" structures which one has on analog tape. It is thus necessary to be able to start the analog tape recorder on a command from the sound processor's digital control system. The possibilities offered by the processor in these respects are enormous, and have previously only been able to be realised in large studios with access to computers.

In order to be able to achieve these more complicated editing procedures, we have to introduce special "instructions" during the recording. The instructions are take in from a fingerboard to a special data box. INSTRUCTION BOX 1016, see page 163. The purpose of the instructions can be said to be the control of functions, which in some way are common for all the soundboxes. It can feed-in data to these from the cassette tape recorder, it can control instructions that are more sound-box orientated. The instructions consist of four figures. The first two denote the type of instruction. The other two can be a number between 1 and 99, which can have a special meaning for certain instructions.

We shall begin with the simplest instruction, which is not actually a real instruction:

OO.XX SET FLAG X

With this instruction, we only put in a numbered "flag" with the number 1-99 on the digital cassette tape. Other instructions use this "flag" to orientate themselves on the digital cassette tape. FLAG 1 is reserved for indicating the beginning of a composition, and FLAG 99, the end. On CONTROL BOX 1009, page 163, we ember that the HOLD MODE knob had the positions FLAG and STOP FLAG. se are used if one wants the reading from a digital cassette-tape to stop each e a flag comes, or only for the final FLAG 99, STOP FLAG.

The next instruction is almost as simple:

XX SET INDEX X

With this instruction, we put in a number between 0 and 99 in a special "index ster". The number in this register can with the help of certain instructions be uced by a whole number. In the next instruction 02.XX CHANGE TAPE, the two numbers are irrelevant. When the instruction reaches the processor from data reader from, shall we say, tape recorder A, the reading continues from e recorder B. The instruction can be used when one wants a greater amount of a than there is room for on one cassette tape, or when one wants readings from cassette tapes alternately.

The remaining instructions in the group OO.XX-09.XX are reserved for future ensions of instructions of a similar character, i.e. settings of index registers, kings, time measurements etc.

The group of instructions 10.XX-19.XX are reserved for instructions concerning sound-boxes. Examples include the phasing in of frequency generators, phasing of certain sound-box groups.

The next group of instructions, 20.XX-39.XX, contain tape movement instructs. A list of these is shown below.

XX GO TO FLAG X BACKWARDS
XX GO TO FLAG X FORWARDS
XX GO TO FLAG X BACKWARDS, IF INDEX> 0, SUB 1
XX GO TO FLAG X FORWARDS, IF INDEX> 0, SUB 1
XX GO TO FLAG X BACKWARDS, CHANGE TAPE
XX GO TO FLAG X FORWARDS, CHANGE TAPE
XX GO TO FLAG X BACKWARDS, IF INDEX> 0, SUB 1, CHANGE TAPE
XX GO TO FLAG X FORWARDS, IF INDEX> 0, SUB 1, CHANGE TAPE
XX GO OUT OF SUBROUTINE
XX GO OUT OF SUBROUTINE, KEEP DATA
XX GO TO FLAG X BACKWARDS SUBROUTINE
XX GO TO FLAG X FORWARDS SUBROUTINE
XX GO TO FLAG X BACKWARDS, IF INDEX> 0, SUB 1, SUBROUTINE
XX GO TO FLAG X FORWARDS, IF INDEX> 0, SUB 1, SUBROUTINE
XX GO TO FLAG X BACKWARDS, CHANGE TAPE, SUBROUTINE
XX GO TO FLAG X FORWARDS, CHANGE TAPE, SUBROUTINE
XX GO TO FLAG X BACKWARDS, IF INDEX> 0, SUB 1, CHANGE TAPE, SUBROUTINE
XX GO TO FLAG X FORWARDS, IF INDEX> 0, SUB 1, CHANGE TAPE, SUBROUTINE

The expression GO TO FLAG X means that the tape is wound forwards or kwards, depending on what is indicated in the instruction until a mark appears, whose number agrees with the X number in the instruction. This jump in quantity of data on the cassette tape can happen unconditionally as for example 20.XX and 21.XX.

Certain conditions can also be given so that a jump is accomplished. In 22.XX example requires that the index register's content shall be greater than nought. If jump is carried out, the index register's content is lessened by 1.

This type of operation is useful when one wants to repeat a structure a cert number of times. For example, should we want to repeat a structure eleven tim we can programme in the following instructions:

01.11 SET INDEX 11
00.05 SET FLAG 05

Structure

22.05 GO TO FLAG 05 if INDEX $>$ 0, SUB 1

The addition of CHANGE TAPE to a jump instruction means that the earl unactive tape recorder is started for reading at the same time that the previou running recorder looks for its mark and then stops there. The instruction makes possible to feed out data from the two cassette recorders for ever without a brea

The addition, SUBROUTINE, to a jump instruction means that the controlli electronic equipment notes that the movement leads to a "sub-structure". With the sub-structure, ordinary jumps can be accomplished. The digital cassette record returns to its position before the subroutine movement as soon as the moveme instruction 28.XX, GO OUT OF SUBROUTINE crops up. The X number is n relevant to this instruction.

The subroutine jump makes it possible to perform a row of sub-structures, e complicated envelopes, which return in several places in an electronic compositi but without the sub-structure needing to be recorded in its entirety.

One can also think of the possibility of preparing a tape with certain pedagogica interesting sub-structures at a larger studio, and then using them for furth composition at the mentioned education centres.

The jump on a digital cassette recorder is naturally not achieved at the speed light. Sometimes, the recorder can wind for several minutes before it finds a ma with the correct number. A jump under such circumstances is completely unusual in a musical context. Fortunately, it is possible to circumvent these disadvantag

Let us consider a long electronic composition, in which we want to make so rearrangements, shortenings and repeats. We then copy in suitable marks and jur instructions. After this we make a further copy. The digital cassette recorder, fr which the data is read, now moves quickly between different places on the ta There are long pauses in the reading during these jumps. However, these are n recorded on the second tape. Nevertheless, we have put the flip-flop connecti above COPY on TAPE COMMANDS 1015 in the position SKIP TAPE INSTRU TIONS, see page 163. This means that we do not copy marks or jump instructio onto the new copy. Thus we obtain a completely pure digital copy without t jump pauses and instructions. This copy is equivalent to any other material an equally be subjected to the addition of new instructions.

We have, finally, a group of instructions which are reserved for the contr e.g. analog tape recorders.

60.XX START TAPE 0
61.XX START TAPE 1
62.XX START TAPE 2
etc.
70.XX STOP TAPE 0
71.XX STOP TAPE 1
72.XX STOP TAPE 2
etc.
80.XX REWIND TAPE 0
81.XX REWIND TAPE 1
82.XX REWIND TAPE 2
etc.
90.XX FAST FORWARD TAPE 0
91.XX FAST FORWARD TAPE 1
92.XX FAST FORWARD TAPE 2
etc.

The X number is irrelevant in these instructions. But one can consider it r gain a meaning if the analog tape could be marked with the same X. Then one e.g. wind forward to a certain X number.

Finally, it should be pointed out that this list of instructions is only prelimi It is proportionally simple to extend and modify it, considering the under electronic control is built.

1.13 Some examples of its use

Ex. 1.

A composer realizes that he obtains an interesting musical effect by sticking contact microphones to an object and taking the signals from these to diff band filters at the same time as allowing another microphone to pick up the s from an oboe he is playing. However, he wants to add a weak signal which co of Radio Luxemburg ring-modulated with the contact microphones and the

Solution: In order to realize this work, he requires:

4 input amplifiers
2 accumulator amplifiers
2 band filters
3 switch boxes
1 ring-modulator
6 attenuators

The connection as in figure 1.13.1 can be used

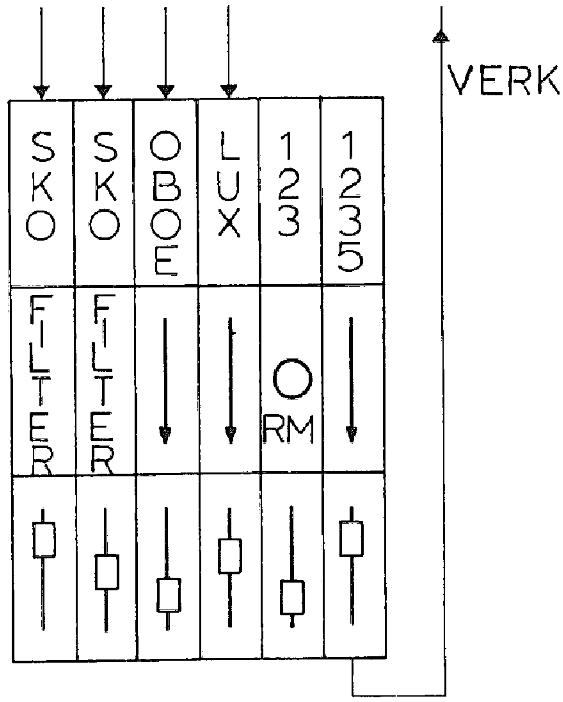


Fig. 1.13.1

Ex. 2

A commercial grammophone company has done a pop-recording on its nev channel tape recorder and has used all 16 channels. The company propos(reduce the recording to 2 channel stereo in order to cut a disc. It is difficult fo: producer to control all 16 channels himself at the same time and he does not c together with his help-mates at the attentuators.

Solution: The producer is able to programme the whole mixing process hin He begins by making a mark with a pencil the beginning of the 16 channel tap(sets in motion a digital cassette recorder (WRITE) and begins by programmin| instruction 61.00 START TAPE 1. The 16 channel tape recorder starts and h(concentrate on the signal level of a few channels. He then repeats the proces: with COPY activated on TAPE COMMANDS BOX. He can now give a level to : more channels by connecting these to LOCAL-MODE. (He pulls out the cas box's upper handle). By repeating this further, he can set all the channels' l himself. He can do as many adjustments as he wants. When he is satisfied, the channel result can be coupled directly the cutting equipment without an i mediate two channel analog tape recorder. The producer saves the 16 channel and the accompanying mix-programme on the digital cassette and treats bo original.

Ex. 3

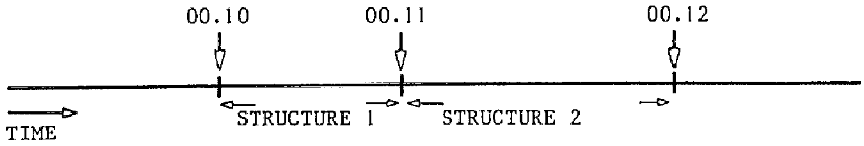
A large theatre performs an operetta, and has several microphones placed on the stage. The actors move between different microphones. The operator of the sound processor has previously put in a ready control programme for the sound-boxes. The programme is fed stepwise from when he presses PLAY, since he has HOLD MODE in the position INSTR (page 163). The prima donna in the performance has however been taken ill and has been replaced by someone with considerably wider vocal dynamics. This causes the overloading indication on the TV-screen to be continually brought into action.

Solution: The operator is able to find out quickly which channel is overloaded. He continues the regulation of signal strength in LOCAL-MODE, and adjusts the level manually.

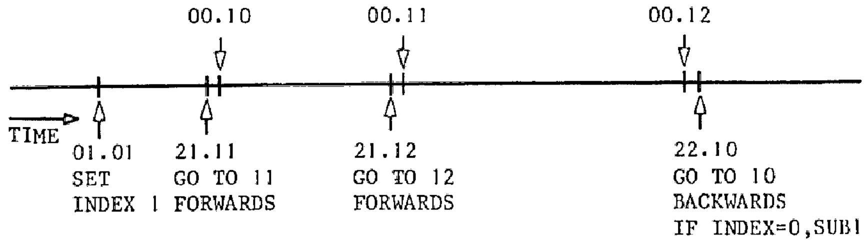
Ex. 4

We shall finish this row of examples of practical use with a simple example of programmed editing. A composer has recorded several electronic sound-structures on a digital cassette tape but now wants to change the position of two of them.

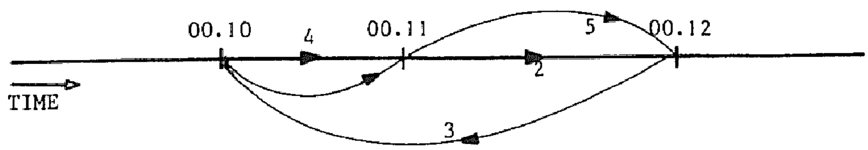
Solution: The composer marks the beginning and end of both structures with a suitable mark-number, e.g. as below.



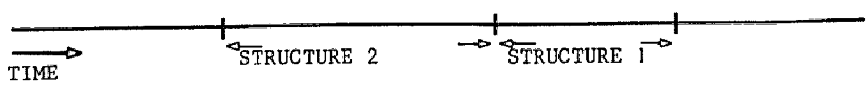
After that he can put in the necessary jump instructions that are required:

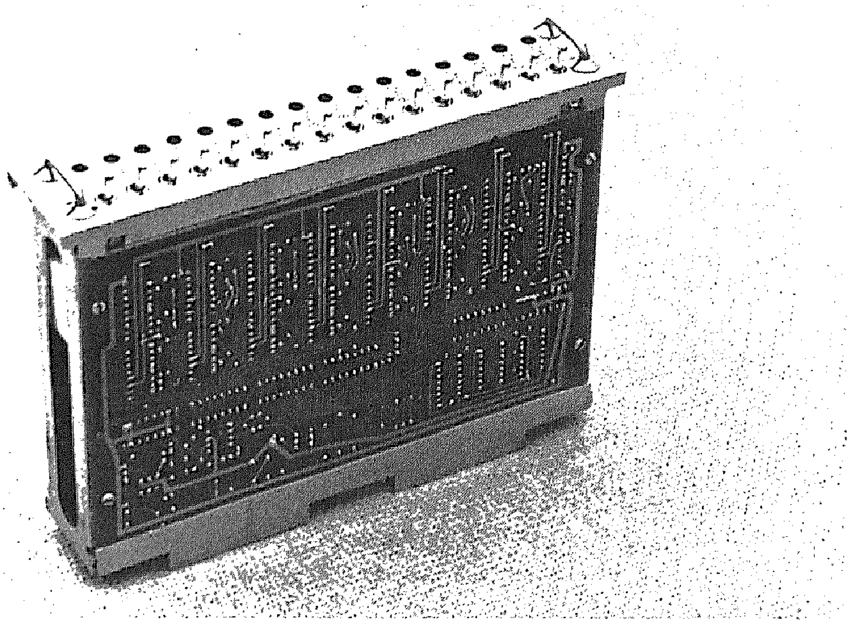


The digital cassette recorder will then move the programme in the following way:

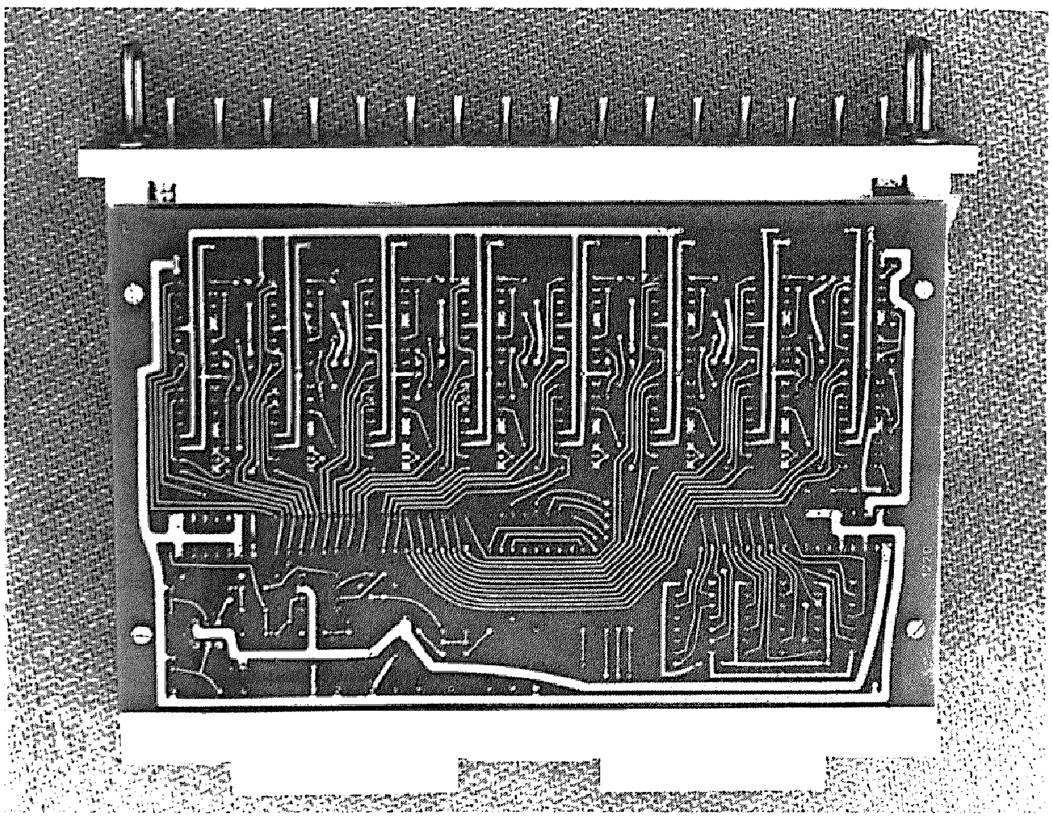


If the sections are copied and the switch is put in the position SKIP TAPE INSTRUCTION on TAPE COMMANDS 1015, the structures then change position and the earlier instructions are erased.

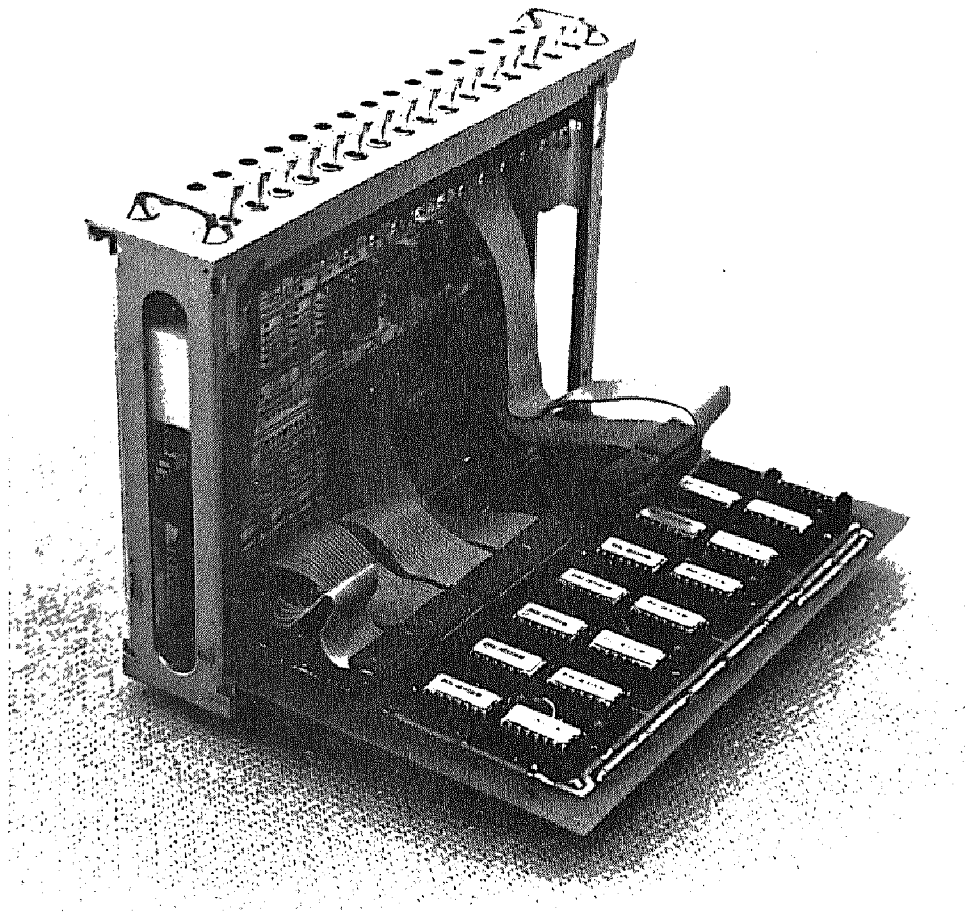




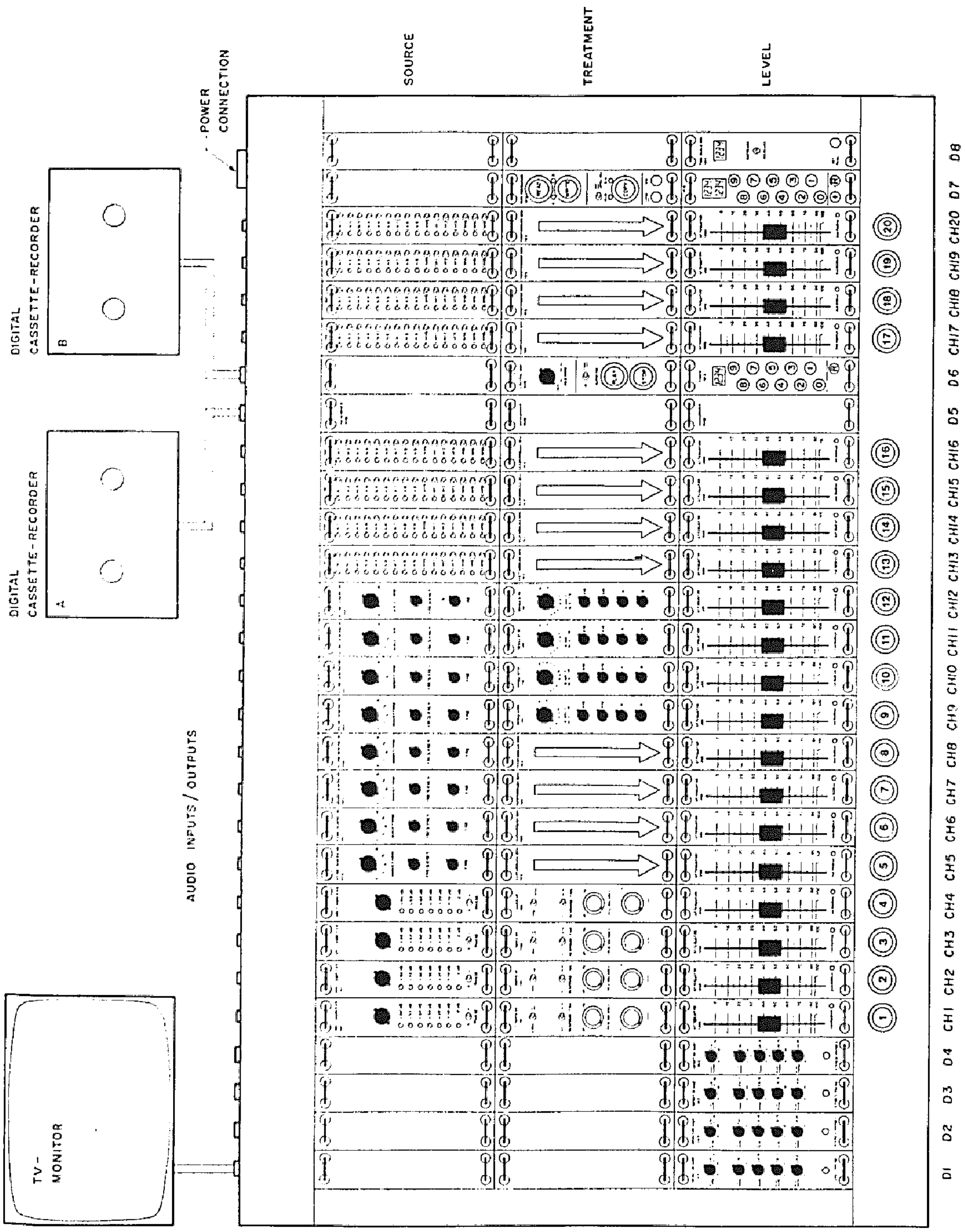
Typical audio-box



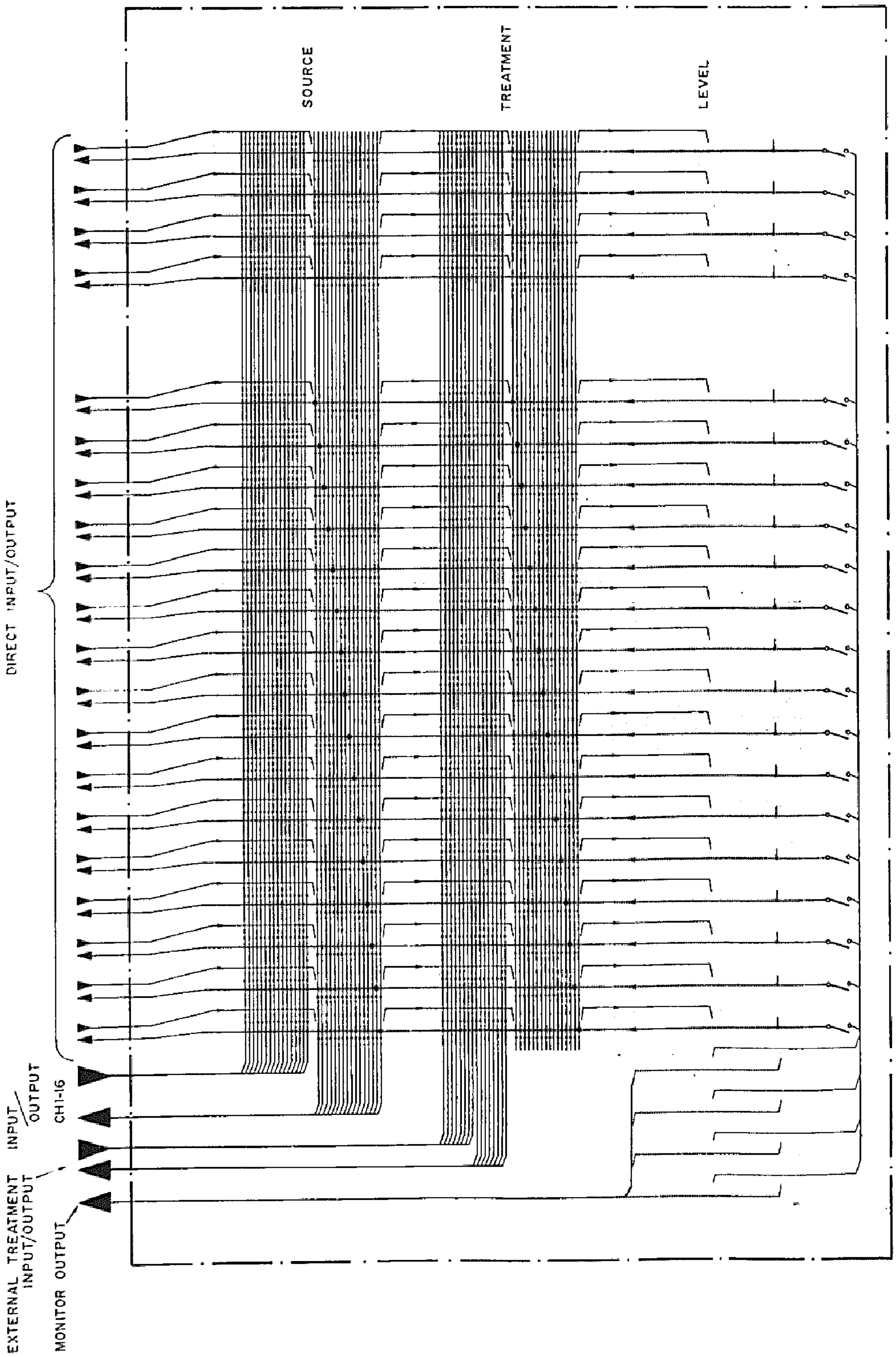
Audio-box with handles in lock and local position



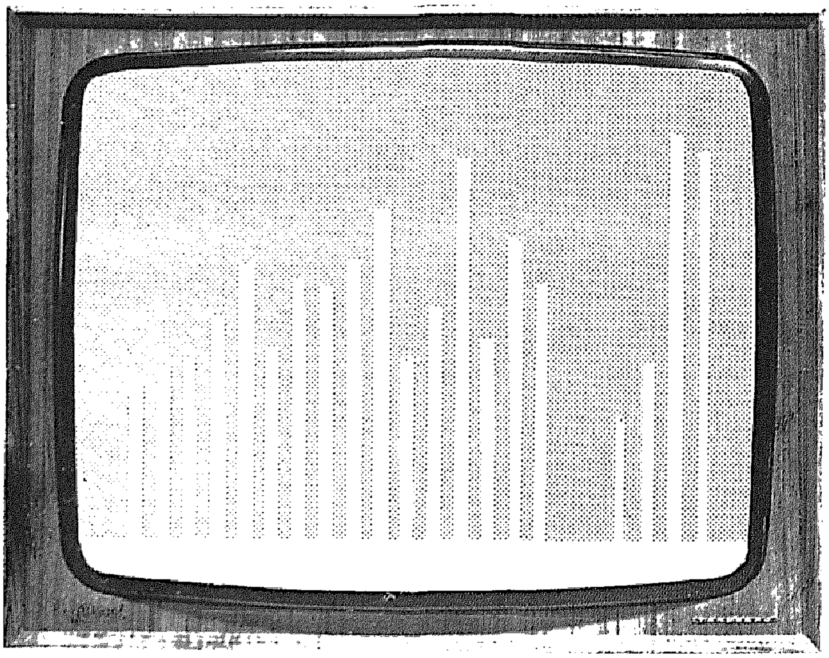
Picture showing a typical audio-box inside



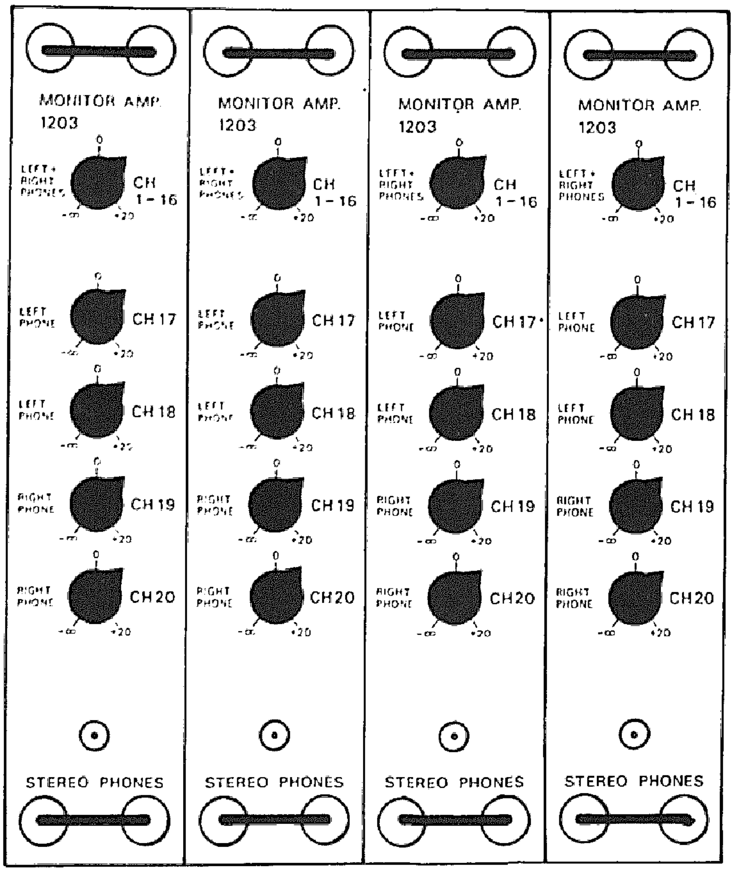
THE PROCESSOR WITH A TYPICAL SET OF AUDIO-BOXES



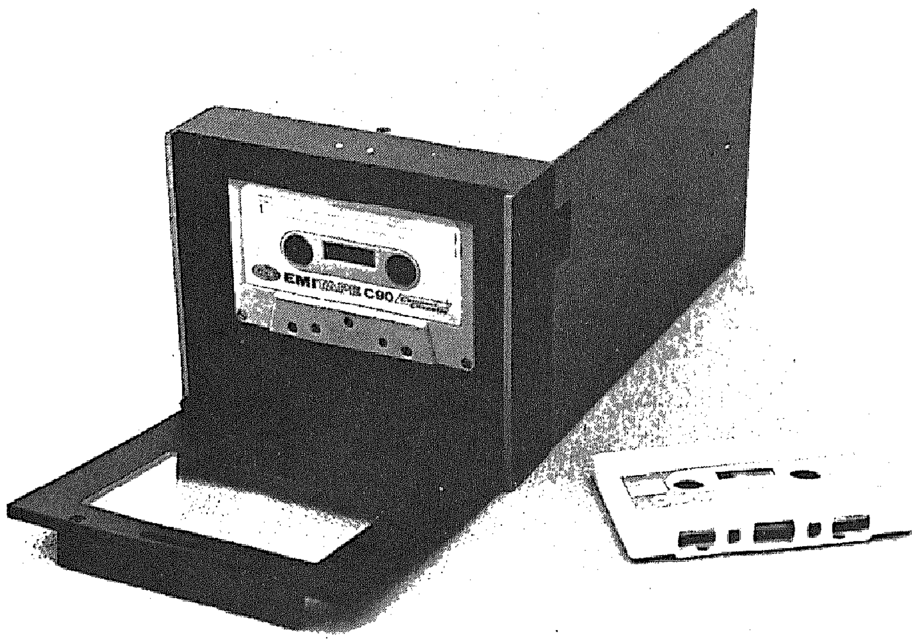
AUDIO-PATHS IN THE PROCESSOR



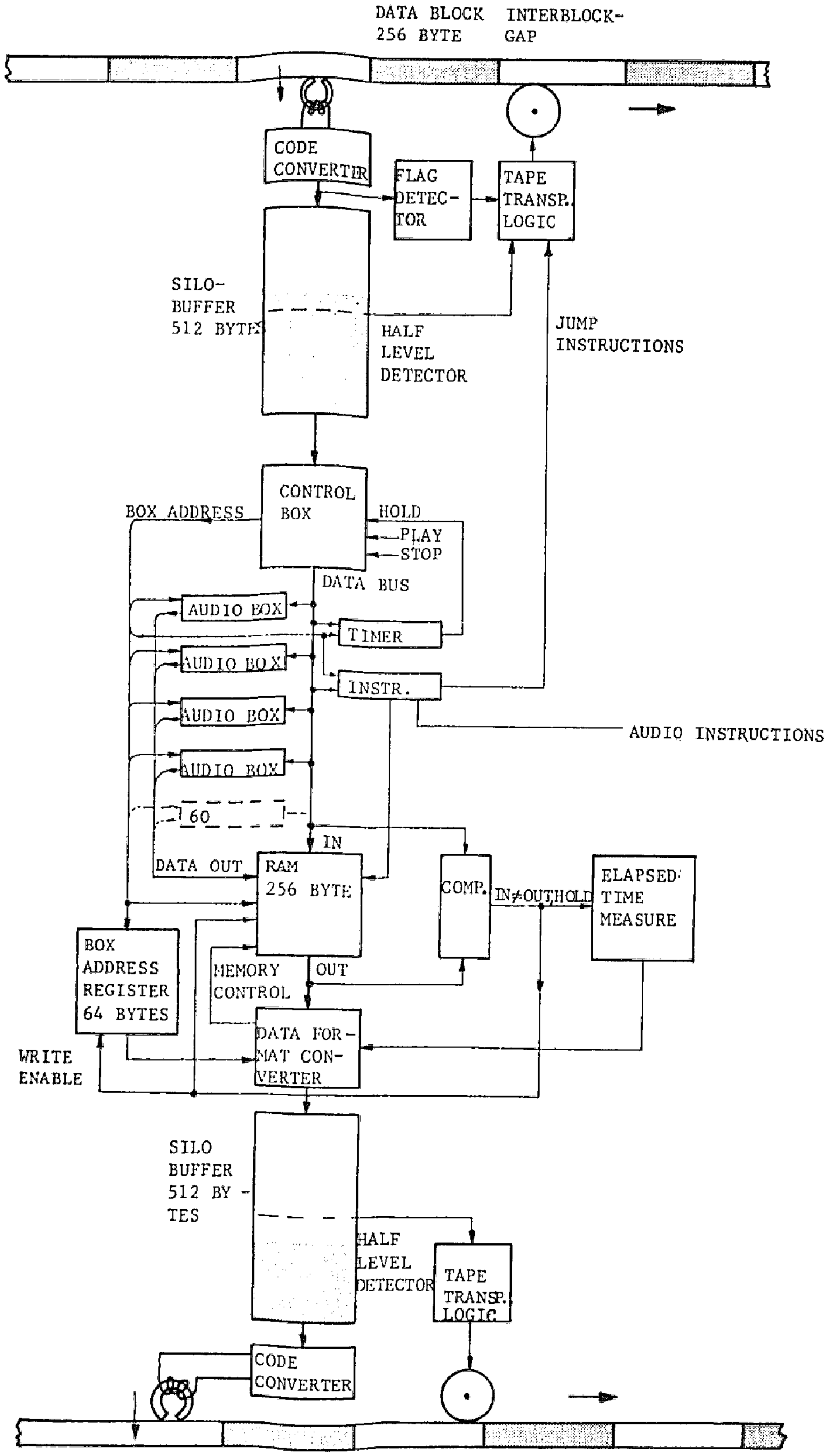
LEVEL-
MONITOR

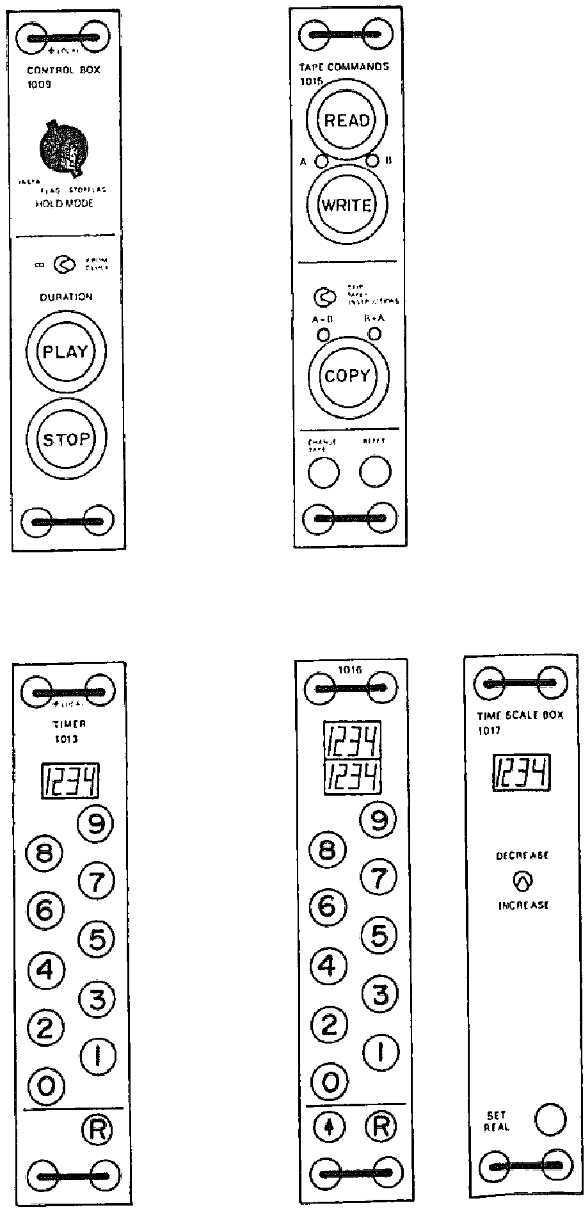


MONITOR
AMPLIFIER

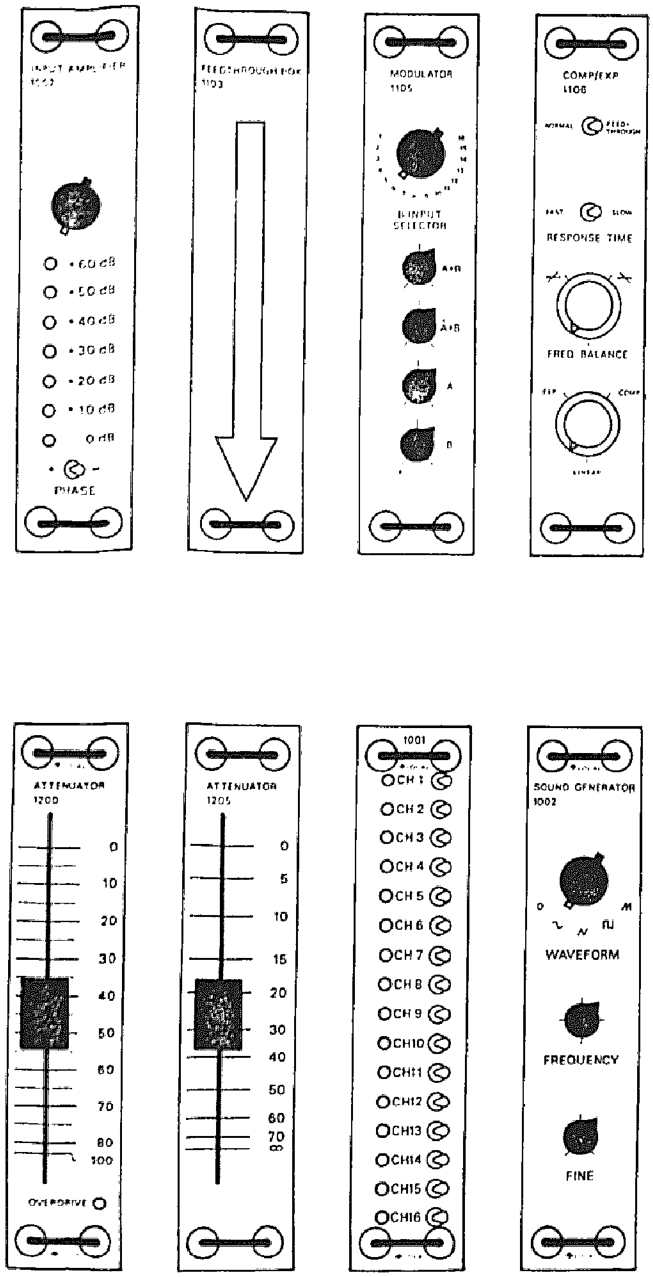


Digital cassetterecorder

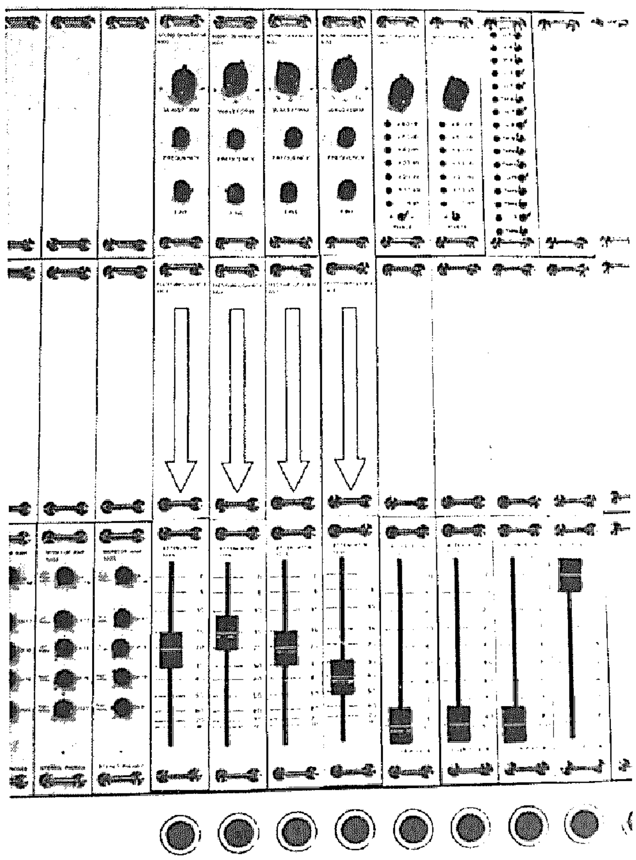

DATA BLOCK
256 BYTE
INTERBLOCK-
GAP
CODE
CONVERTER
FLAG
DETEC-
TOR
TAPE
TRANSP.
LOGIC
SILO-
BUFFER
512 BYTE
HALF
LEVEL
DETECTOR
JUMP
INSTRUCTIONS
CONTROL
BOX
BOX ADDRESS
HOLD
PLAY
STOP
DATA BUS
AUDIO BOX
TIMER
AUDIO BOX
INSTR.
AUDIO BOX
AUDIO INSTRUCTIONS
AUDIO BOX
60
IN
DATA OUT
RAM
256 BYTE
COMP.
IN≠OUT
HOLD
ELAPSED
TIME
MEASURE
BOX
ADDRESS
REGISTER
64 BYTES
MEMORY
CONTROL
OUT
DATA FOR-
MAT CON-
VERTER
WRITE
ENABLE
SILO
BUFFER
512 BY-
TES
HALF
LEVEL
DETECTOR
TAPE
TRANSP.
LOGIC
CODE
CONVERTER


CONTROL BOX
1009
HOLD MODE
DURATION
PLAY
STOP
TAPE COMMANDS
1015
READ
A
B
WRITE
COPY
TIMER
1013
1234
9
8
7
6
5
4
3
2
1
0
R
1016
1234
1234
TIME SCALE BOX
1017
1234
DECREASE
INCREASE
SET
REAL

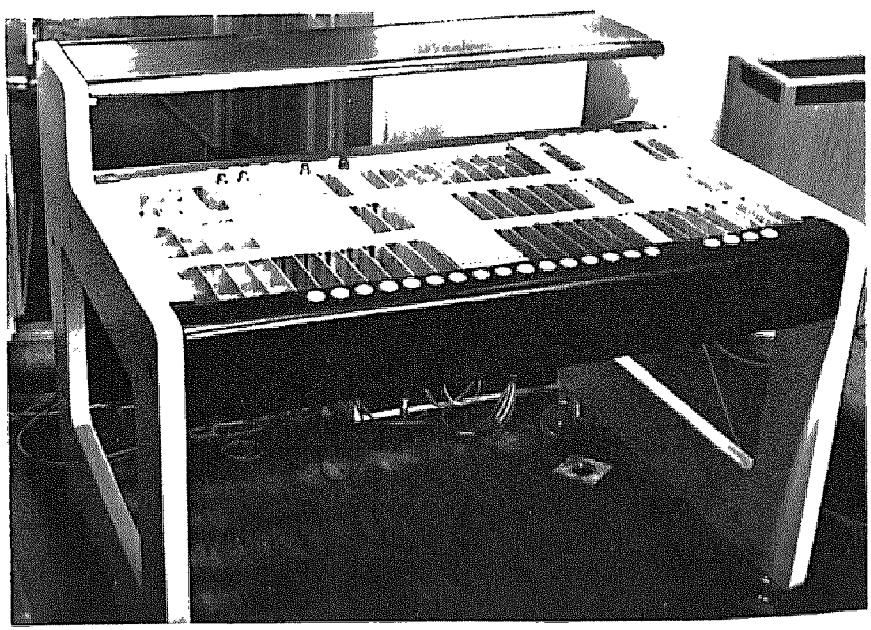
DATA-BOXES


MODULATOR
1105
COMP/EXP
1106
NORMAL
FEED-THROUGH
FAST
SLOW
RESPONSE TIME
FREQ BALANCE
COMP
+60 dB
+50 dB
+40 dB
+30 dB
+20 dB
+10 dB
0 dB
PHASE
A+B
A
B
ATTENUATOR
1200
ATTENUATOR
1205
OVERDRIVE
1001
CH 1
CH 2
CH 3
CH 4
CH 5
CH 6
CH 7
CH 8
CH 9
CH10
CH11
CH12
CH13
CH14
CH15
CH16
SOUND GENERATOR
1002
WAVEFORM
FREQUENCY
FINE

AUDIO–BOXES



A closeup picture of the processor



Picture showing the first prototype of the processor