COMPUTER-AIDED COMPOSITION AND PERFORMANCE WITH AMUS

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ABSTRACT

An exciting new micro-computer-based sound generation system has been developed at North Texas State University (NTSU). The Automatic Music System (AMUS), designed and developed at NTSU, has found uses in music instruction and in composition. One system has been in use since August, 1977. As it was an economical and successful instrument, the school of music is building an additional ten systems to be put into use on the NTSU campus for a variety of musical purposes. This paper describes the nature of the AMUS hardware and the results of using it to develop and perform a specific musical composition, *Monodram*. AMUS provides the composer with an economical tool which frees him from a significant amount of tedium, and allows him to spend more time in the creative aspects of the compositional art.

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INTRODUCTION:

An exciting new micro-computer-based sound generation system has been developed at North Texas State University (NTSU). The Automatic Music System (AMUS), designed and developed at NTSU, has found uses in music instruction and in composition. One system has been in use since August, 1977. As it was an economical and successful instrument, the school of music is building an additional 10 systems to be put into use on the NTSU campus for a variety of musical purposes. This paper describes the nature of the AMUS hardware and the results of using it to develop and perform a specific musical composition, *Monodram*. AMUS provides the composer with an economical tool which frees him from a significant amount of tedium, and allows him to spend more time in the creative aspects of the compositional art.

DESIGN CONCEPTS:

Several important design considerations influenced the development of AMUS. The most important of these was the desire to find a viable middle ground between "no holds barred" systems, very elaborate but too expensive, and "toys", very affordable but too limited. AMUS is moderately affordable and moderately sophisticated. In the AMUS system the hardware sound generator (called Musor) handles the details of performance, i.e., the actual generation of analog voltages by means of digital-logic techniques. A microcomputer (the Motorola M6800) controls the sound on a macro level, determining the timing, pitches, amplitudes, and envelopes of the notes during performance. The tones produced by Musor have predetermined (fixed) qualities of harmonic content, tremolo and vibrato. The microprocessor also accepts scores which use the AMUS notation and translates the information into the format required by Musor.

Musor contains a maximum of eight independent voices, each with an individually definable waveform. This waveform, which may be different for each voice, is fixed by the instrument maker, being determined by the values of 16 resistors (a different group for each voice); the harmonic content is determined by the selection of values for these components. This hardware design decision limits the available waveform characteristics, but not necessarily the available timbres. It also makes possible considerable economies in hardware construction.

The design of Musor allows enough flexibility to serve a variety of users; it can be configured with one to eight voices and a wide range of optional control features. This approach allows an economical minimum configuration to be used for instruction in ear training, and a more expensive maximum performance version, with simple changes in hardware, to be used for other applications with the same microprocessor and software.

The Musor hardware implementation also takes advantage of design idiosyncrasies, i.e., features which are easily and inexpensively added because of the nature of the hardware. For example there is a feature which mirrors two quarters of each cycle of the waveform, to produce added "brightness" (a waveform modification which results in sharp edges). Implemented by the addition of one integrated circuit per voice, this feature creates a very different waveform and timbre, under control of a manually-operated switch or under control of the score. Other examples are the features permitting display of the measure number currently being performed, and the control of non-musical external devices by means of the musical score.

A final important consideration in AMUS design was the development of a versatile, readable, user-oriented score language. It must be possible for the score information to be entered, displayed, edited, stored, or created by a wide variety of terminals and computer systems.

In particular the use of cheap terminals and standard BASIC must be practical. In the models of AMUS so far constructed a Cathode-Ray-Tube (CRT) terminal is used, together with a 32-port Hewlett-Packard model HP-2000 BASIC timesharing system. For this reason the AMUS score language uses ASCII alphanumeric characters.

The language is best suited for multiple-part note-oriented music using the equal-tempered scale and conventional rhythmic values. However the user can define alternate tunings with accuracy or work at a lower level and specify pitch parameters directly. The score language is thought to be one of the strengths of the system, as it allows a great deal of detailed control when desired, but it also provides a large number of automatic default values to simplify encoding. When different values from the default values are necessary, the user may give the value for an individual note or give the value which is to apply to all the following notes in the

AMUS score. Global values for articulation, scale, transposition, and metronome markings, for example, allow simple coding and easy modification. (See Hamilton and Scott, 1978, for a detailed description.[1])

THE COMPUTER SYSTEM.

A Southwest Technical Products (SWTP) M6800 microcomputer was chosen as the processor for AMUS. The low cost of the M6800, its ability to respond quickly and flexibly to the needs of peripheral devices, its easy input and output instructions, and in general its good instruction set made it the best choice in its class of processors for this application. The Musor instrument is accessed as if it were a location in memory, making for a simple hardware interface.

The software, contained in four thousand bytes of Read-Only-Memory, is primarily devoted to score translation. The input (which can be entered at a terminal keyboard, from a tape cassette, or over a phone line that is connected to another computer) is translated from ASCII standard characters into an internal score containing all of the information needed by Musor. The score information is stored in variable-length blocks in the following format:

Type	Length	Contents
Header	1 byte	Length of block, 16-bit (2-byte) words.
	1 byte	Measure number.
Note data	1 byte	Periodicity (Musor pitch information).
	1 byte	VVVVAAOO, where VVVV is 4-bit voice/register, AA is the amplitude code, OO is the octave code.
Trailer	2 bytes	Timing (duration).

Two bytes of note data are needed for each voice which changes at the beginning of the time interval.

In playing a piece the program interprets the data from each block in the manner illustrated by figure 1.

The duration timer is an interrupt-driven program which counts down in units of 568 microseconds until the 16-bit count, set by the playing program, reaches zero. At this point the next block of internal score information is sent to Musor, and the timer is set to the new duration value. Performance can be concurrent with input and translation processes, allowing the beginning of a score to be played while the remainder is still being read and translated.

A note produced by Musor is composed of as many as four successive note fragments. A note fragment is defined as a musical sound having duration, constant pitch and an envelope determined by one amplitude. The data for each fragment contains information for one or more voices. The four fragments are analogous to the attack (two fragments), steady state, and final decay of the envelope of a note. Including rest (silence), four constant amplitude levels are possible, plus a fifth amplitude level which decays over time. Figure 2 illustrates a typical envelope. Scoring conventions make it possible to define the amplitude of each fragment separately for each voice, or to use any one of several preset articulation fragments. The durations of each may also be controlled through the score notation (although not differently for each voice).

The translator also handles key signatures, accidentals, transpositions, metronome markings, and a number of convenience features, such as keeping track of measure numbers to allow the initiation of performance at a given bar.

Finally, the software provides differing modes of interaction so that the AMUS system can be used in conversational applications as an intelligent

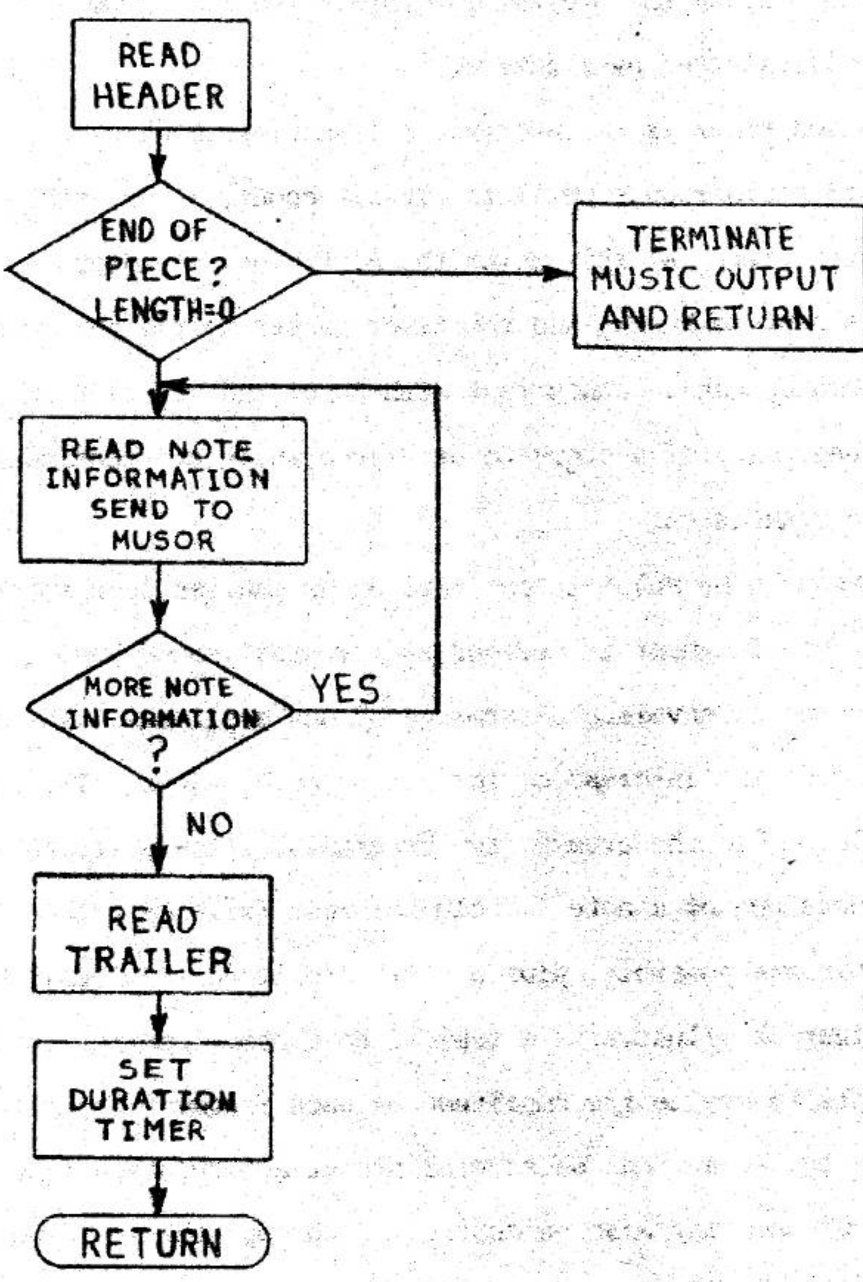

Figure 1. Process for performance of translated score.

terminal. The basic interaction mode is the text mode; in this mode of operation text received from the control terminal is sent transparently to the remote computer, and vice versa, until a special character sequence, beginning with "%", is input. The character which follows the percent sign may signal a number of events, including the transition from text mode to a score-translation mode. In the score mode all incoming text is translated, as a score in the AMUS notation, but normally is not displayed. This feature allows both visual and aural communication.

MUSOR, THE INSTRUMENT.

The Musor instrument consists physically of a card cage, edge connectors, wiring assembly, 5-volt and plus- minus- 15-volt power supplies and a control panel. It is connected to the microprocessor computer system by a short cable. There are five types of 6-1/2 inch printed-circuit boards: I (computer interface), D (digital-signal clock generation), A (audio generation from digital signals), R (control registers), and M (audio mixer and output amplifier). A minimum configuration includes one each of boards I, D, A, and M. The maximum model has one I, two D, eight A, two R, and one M, for a total of fourteen boards. The R (control register) boards are optional; their function may be performed by manual controls. Figure 3 is a block diagram of the complete instrument.

The Musor subsystem contains up to eight individual waveform generators, the outputs of which are mixed into a single analog (voltage) output. Each generator is of the same design, but the waveforms may be defined differently for each voice, as may other aspects, such as pitch range and rate of envelope decay. The final output can be connected to headphones, to an amplifier, or to a recorder.

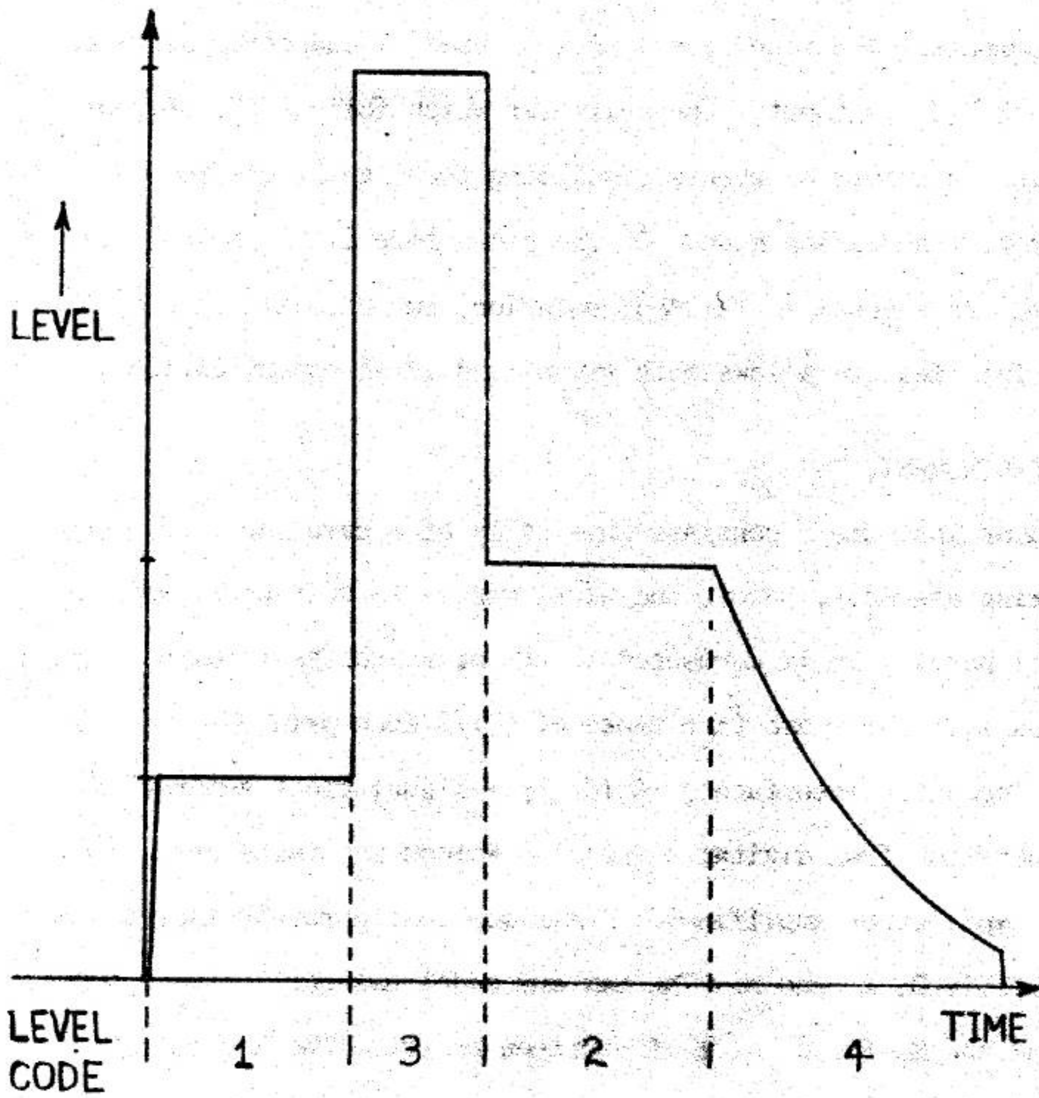

Figure 2. A typical Musor note with an envelope consisting of four note fragments. Each of the four level codes is illustrated in this example. Note that the fragments are of variable length and, in fact, can have durations determined in the score.

Pitch. The fundamental period of the audio signal produced by a Musor voice is perceived as pitch; it is determined during performance by the values for two integers supplied by the microprocessor, the octave and the periodicity codes.

The audio signal of Musor produced by board A has 64 discrete amplitude values per cycle. The pitch is determined by the rate at which Musor steps through these amplitude values. The digital clock signal which accomplishes this stepping for each board A is created by a circuit on board D; it divides a 17,000,000 hertz master oscillator frequency by a value which is determined by the octave and periodicity codes for each note fragment. Figure 4 shows the logic of the divider circuit.

The octave code is a two-bit value and the periodicity is an 8-bit value (usually) chosen from a table stored in the microprocessor memory. Periodicity values for a well-tempered scale are provided; or the use can define his own table values of up to 12 tones per octave, or he can set the values individually for each note fragment. The range of a voice is 5 octaves, with average pitch accuracy of 3 cents (9 cents error for some pitches in the highest octave).

Timbre, as used below, refers to the waveform of a given Musor voice. Board A of each Musor voice produces 64 amplitude values during one cycle; there are, however, only 16 different (independent) amplitude values. These are determined by the values of 16 resistors used as parts of a digital-to-analog converter. The resistors define 16 amplitude levels for each quarter of a cycle. Normally the same 16 amplitude values are generated again, in reverse order, during the next quarter cycle (i.e.,

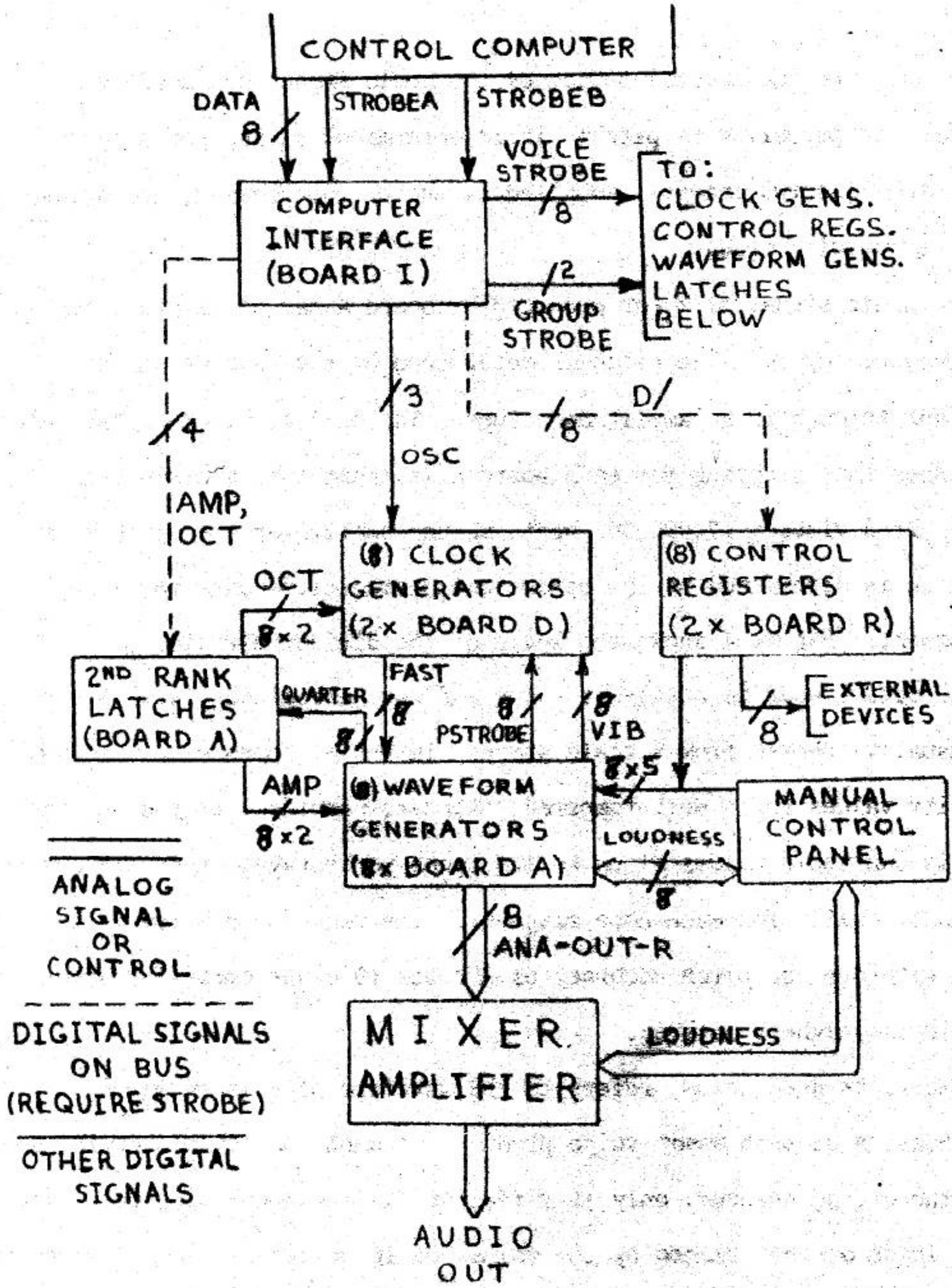

Figure 3. Block diagram of the Musor instrument.

as 0 1 2 3 ... 13 14 15 15 14 13 ... 3 2 1 0), to form a half cycle of only positive amplitudes. This digital sequence, and hence, amplitude sequence, repeats indefinitely, until the pitch, octave, or amplitude code is changed.

The always-positive signal is fed into a unity-gain amplifier which has its gain reversed by a digital signal every half cycle in time, alternating in gain between +1 and -1 (approximately). This action produces a negative-polarity mirror image of the first half cycle of the waveform during the second half cycle.

Thus a 64-step bipolar cycle is produced with 16 independent amplitude values. Since the waveform has quarter-wave symmetry, only odd harmonics can be produced by this method. The resistor values are chosen to give the desired proportions of odd harmonics.

By designing the unity-gain amplifier to switch gain between +1 and about -0.8, instead of from +1 to -1, even harmonics are introduced, adding to the complexity of the waveform.

Vibrato is produced by means of a low-frequency oscillator provided for each voice; its digital signal is used to perturb dynamically the digital periodicity parameter used in frequency division.

The audio signal is sent through a low pass filter and then is mixed (board M) with the signals of the other voices to produce a single output. Various voltage and power levels for output are provided.

Amplitude. The amplitude of the waveform envelope for a voice is controlled in two ways. One feature provides for the fast changes in envelope which are needed for articulation; the other provides for a relatively large jump in loudness, which can be applied only between notes (due to the possibility of generation of non-musical clicks).

The first method associates one of five level code values with each note fragment. Musor level codes 1, 2, and 3 imply amplitude steps differing

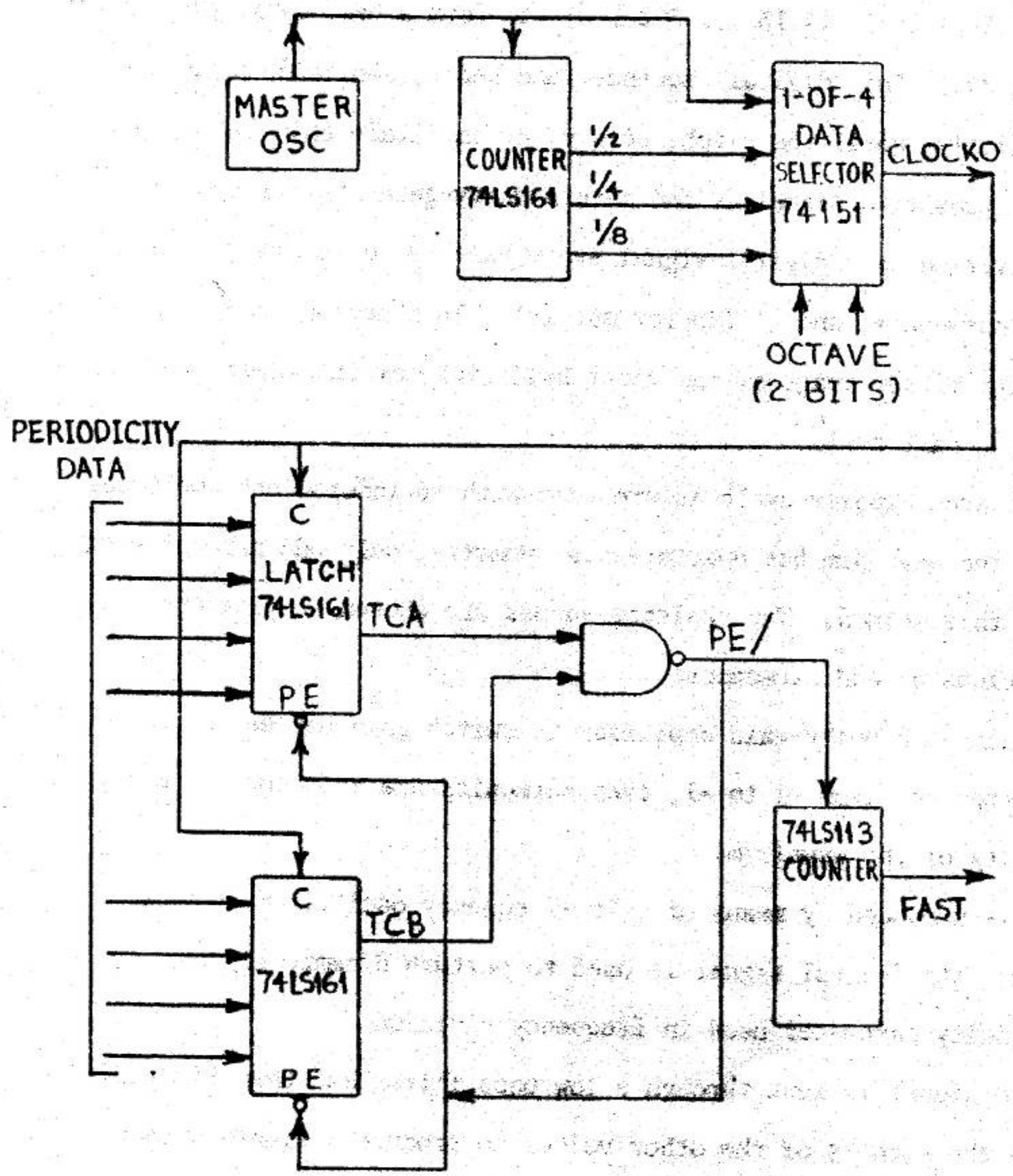

Figure 4. Logic design for frequency divider. (Some timing details have been omitted in order to simplify the diagram.)

TCA & TCB = Terminal count (count = 1111)
PE = Parallel entry enable
FAST = Final divided output, 64 times the desired frequency.
CLOCKO = Oscillator output divided by the octave designator.

by about eight decibels. The translator causes rests to have amplitude level 1, but with a very high frequency, which is rendered inaudible by filtering.

AMUS score level code 4 is translated to Musor code 0, which implies an eight-step decaying envelope (starting with the same level as code 2). The amplitude decays in discrete steps of about 1.7 decibels, as determined by eight resistors. The last amplitude value is zero. The rate at which the decay occurs is proportional to the pitch, thereby following the characteristics of many other instruments. A "decrementing parameter" ranges in value from 0(no decay) to 7 (maximum rate of decay); this parameter selects how many cycles of the waveform must occur before one step of decrementing occurs. The value of the decrementing parameter is determined by a register or by a manual switch (for each voice). The aural effect ranges from no decay at all to an almost immediate cutoff.

The second method of amplitude control provides for 12 decibels of immediate change. A control register can be provided to set one of two amplitude levels for each voice; alternatively, a manually operated volume control (potentiometer) can be provided.

Tremolo is generated by using the digital signal from the vibrato oscillator in such a way as to vary periodically the amplitude of the positive half-cycle of the audio signal. Thus, the harmonic content of the waveform is also perturbed, with a complex result.

The effects of changes in amplitude, pitch, and octave codes are delayed by Musor after receipt from the microprocessor computer; the changes take place only as the waveform passes through zero value, so as to minimize audible clicks caused by waveform discontinuities.

A COMPOSITIONAL APPLICATION OF AMUS.

The composition *Monodram*,[3] is the first original work to use the AMUS system extensively. It is the combination of both computer-assisted composition and computer-generated performance using the MUSOR instrument. *Monodram* is designed to stand alone as a composition for tape, or it may be presented with a Player. The Player performs a theatrical song/presentation of the poem "Monodram", by Rosanne Dill.[4]

Monodram is formed based on principles of Golden Mean arithmetic.[5] The entire duration is multiplied by .618 to determine the primary point of highest activity. The durations of the two resulting sections are then each multiplied by .618 in order to determine secondary strong points. This process continues until the work is divided into twenty-two modules. Each module is a self-contained entity having a duration of from three to thirty seconds. (See figure 5.) Each is a modified arch, in which areas of greatest density are arranged in relation to the Golden Mean. The density of events in each module is defined on a scale of one to five, with one being the least dense. Elements determining the density include the number of events per module, their relative speed, and their volume. The composed elements of one module reappear in other modules, allowing for motivic development. These elements are mutated by transposition, truncation, expansion, inversion, or other compositional devices.

A BASIC program was written by the composer to assist in composing the modules in *Monodram*. The program chooses, at random, the duration of each module, computes the Golden Mean, chooses a relative density, and suggests one of four compositional techniques:

1) Free stochastic twelve-note composition, in which the tonal vocabulary consists of the traditional chromatic scale tones. The choice of the tones, individual note durations (in the range of 64th to whole notes), voice number (in the range of one to six), and octave register (in the range of two to six, using the American Acoustical Standard), is accomplished by the BASIC program.

2) Free stochastic microtonal composition, in which the pitch is chosen at random, as well as the duration, voice number, and octave register, by the BASIC program.

3) Use of original music by the composer, in which elements of computer-generated music are interpolated within the new music.

4) Use of quotations from extant compositions from a variety of styles and periods, chosen by the composer and transcribed for MUSOR.

Compositional Technique: original music.
Relative Density: 1.
Duration: eight seconds.
Golden Mean: five seconds.
Player: not present.

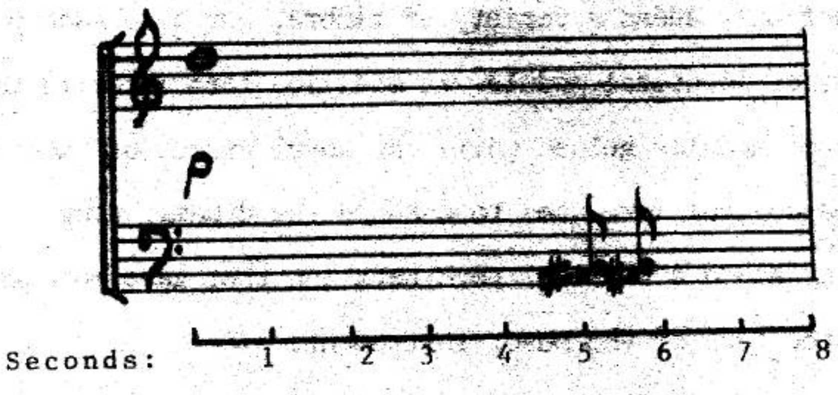

Figure 5. A representative module.

The program also suggests the presence or absence of the Player within the module. When the program selects one of the stochastic methods, it proceeds to print a score in the AMUS language. Files are used to store the score in order to facilitate rapid access at a later date. This score generation is particularly helpful when the microtonal technique is selected as it eliminates the time consuming process of calculating the numerical values for periodicity and octave parameters. The suggestions for each module are played by the Musor instrument, permitting the composer to alter, retain, or reject the module.

In the composition of *Monodram* the composer exercised all of these options. After several hundred modules were generated by the program, the composer edited and combined them to create the written score. The modules consisting of original music and altered excerpts were encoded for Musor manually. Final checks and alterations were made on the AMUS score, and an audio tape was recorded.

In order to overcome some of the limitations imposed by the instrument, the composer employed some manipulation of the audio tape. Since the Musor instrument has a relatively low variety of available timbres, some modules went through processes of ring and frequency modulation. This modulation not only added a variety of timbre, but also made possible the use of similar or identical modules in modified form later in the work. Such reuse of modules helped unify the aural effect of *Monodram*. Panning and volume control were used to enhance the dynamic range. Finally, tape delay added a physical reverberation that gave more warmth and resonance to the tone of the Musor instrument.

A COMPOSER'S EVALUATION OF AMUS

The Musor instrument and the AMUS language offer at least three advantages to the composer interested in working with computers. First, the accessibility of the system is exceptional. The Musor instrument provides an economical way for interested persons to have digital synthesis at their control for many musical applications and experiments.

Second, the AMUS system helps satisfy the need for immediate aural feedback. This real time feedback greatly increases the speed and efficiency of composing. The time consuming processes of writing, copying, procuring the services of instrumentalists, or taping , splicing, and editing are significantly reduced. Using sound-generating digital hardware, the composer immediately decides on which musical excerpts to store, purge, or modify. If he decides to keep any portions not immediately useable, they may be stored for later retrieval.

A third advantage lies in the fact that scores can be generated directly through the use of the widely available BASIC language. The role of composer-written programs can range from making suggestions to generating a complete score.

In comparison to analog synthesizers, such as the large Moogs, Buchlas, or ARPs, the Musor instrument has limitations. The specification of microtones is rather awkward, as the software is designed only for a twelve-tone octave. The waveform of a given voice is not changeable without revision of the hardware; therefore the variety of timbre is not satisfactory. The level of volume control through the AMUS language is not adequate.

In the compostion of *Monodram* the composer realized a step in the direction of composer-computer interaction on a more personal level than has previously been possible, except with the most sophisticated hardware. The amount of time spent in tape manipulation is more than accounted for in the saving of time from clerical tasks. The Musor instrument in combination with the AMUS language is a tool which can free the composer from a significant amount of tedium. Such systems allow the composer to spend more time in the creative aspects of the compositional art.

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