COMPUTER TECHNOLOGY

A computer-controlled system for the recording modification and presentation of two-channel musical stimuli

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This paper describes a computer-based system for the on-line experimental use of musical stimuli. The system is in two sections. The first records in digital form two successive monophonic keyboard performances and combines them into a two-part musical stimulus consisting of bass and treble melodic lines. This musical stimulus can be modified in one or more ways by means of an editing program. The second part of the system presents the modified stimulus to the subject and records his reaction time to particular target notes. The system can be expanded to allow the merging of a larger number of melodic lines into a single polyphonic form or can be modified to produce precisely controlled two-channel tonal stimuli.

A number of psychological experiments involve the subject in listening to more than one stimulus at the same time while responding to targets within the stimulus material (Kahneman, 1973). A series of experiments carried out by the author required the subject to listen to a musical sequence of notes and to respond with a keypress when an unexpected timbre change (from sine to square waveform notes) occurred. The subject's reaction time to these timbre changes was taken as an index of his processing capacity.

The system is based on a LINC-8 computer to which is interface a music keyboard for input and voltage-controlled oscillators for output of the melodic lines. A keyboard performer plays the piece of music in two parts, the treble and the bass parts being played successively. The first part is replayed to the performer while he plays the second part. The input program then combines these two parts and stores them on a computer system device ready for editing and replay to the subjects. The editing programs provide the facility of altering the timbre in which the melodic line is played. The replay program, besides presenting the stimulus to the subject, records the latency of his keypresses in response to the target events in the musical stream, and also which key was pressed.

To meet the requirement that the target events should not be expected, the changes in timbre were programmed to occur at randomly distributed points in time. An auxiliary program generates a piece of pseudomusic from the original input, which comprises notes of randomly distributed pitch having the same duration as the notes of the musical

stimulus. These last two requirements of the stimulus material necessitated the use of a computer system. It would be impractical under normal conditions of the psychology laboratory to arrange for musical stimuli which could be precisely controlled in this way. An analogue system of sound recording, storage, and reproduction would raise problems of detecting the onset of target events and hence measuring response latencies with any accuracy. The system described here, once created, has the potential for generating a wide range of stimuli for this type of experiment.

HARDWARE

The hardware used is illustrated in Figure 1. The experimenter communicates with the 8 K PDP-8 computer and LINC subsystems via a teletype; LINC tapes and DF32 magnetic disk store provide the system-storage devices.

The performer generates the music by playing on a four-octave keyboard purchased from D. E. W., Ltd. Contact closures caused by key depressions are detected by bit positions in four input registers of a Digital Equipment M735. The keyboard has, in fact, 49 notes (the extra note is used for control purposes and is connected to a sense line in the LINC).

Output from the computer is via Analogue Devices 10-bit digital to analogue converters (DACs), Model DAC 10Z/3, fed from the output registers of the M735.

Two channels of sound synthesis were implemented in the system. Two matched and tracked oscillators, D. E. W. Model VCO2, are driven from the DACs. The VCO2s provide phase locked sine or square waveforms from two independent outputs, and it was arranged to switch between these by means of a LINC relay on one channel. The waveform is next passed to a volume control D. E. W. Model VCA1 which cut

Figure 1. System hardware.

back the signal using 4 bits of an M735 output register on each channel. The resulting signal is fed to an envelope shaper D. E. W. Model ES2 triggered by LINC relay, and the two channels are directed to a Leak Stereo 20 amplifier. Presentation to the subject is from two Wharfedale loudspeakers or Koss ESP9 electrostatic stereophones.

The subjects responses consist of keypresses on the music keyboard, detected by the appropriate bit pattern in the M735 input registers.

Timing of the program operations on files and keyboard is done using the KW8/IE programmable real-time clock.

SOFTWARE

The system comprises a music write program (MUSICW), which accepts notes from the keyboard

performance and translates them into stored music, a music play program (MUSICP) which plays the finished piece to the subject and measures his reaction times, and a number of file handling programs, including the editing program.

MUSICW and MUSICP were written in PAL8 assembly language and the other programs were written in BASIC. All programs run under the control of OS/8 and return to the OS/8 system on CTRL/C interrupt.

MUSICW

The music write program operates on keyboard input in two passes, corresponding to the two parts of the piece of music being performed. In the series of experiments referred to earlier, these two parts were the treble melody first followed by the bass accompaniment.

During Pass 1 of the program, the incoming stream of notes and rests is read and stored as indicated in Figure 2. Each note is stored in a four-word unit (quartet). The first word is blank; the second contains the frequency of the note; the third is the duration of the note in milliseconds; and the fourth word contains information in packed binary format for the control of the volume and timbre of the note.

During Pass 2, the notes stored during Pass 1 are re-read and loaded into the clock, where they are used as a time base. Notes played on the keyboard are compared with them, and the music is divided up into time slices, which are defined in duration by the length of time for which neither of the notes changes. These slices are stored in quartets. The first word of the quartet contains the frequency of the note to be played on Channel 1. The second word contains the Channel 2 frequency. The third word is the duration of the time slice. The fourth word contains the volumes of the notes and the timbre of the Channel 1 note.

Figure 3(a) is the flowchart of the program MUSICW. In Pass 1, after initialization, the clock is loaded with 4095. This defines a time interval of 4.095 sec, since the clock count is incremented every millisecond. The keyboard is read and tested for a new note (or rest) occurring, until such an event takes place. At any time the pass may be terminated by pressing the control key. When a new note is struck, the current note is written to storage and the keyboard is read again. The note lengths are calculated from the value of the real-time clock. The clock causes a system interrupt when the count is exhausted, and in Pass 1 the interrupt service routine then resets the clock to 4095.¹

During Pass 2, the timing of the slices is done by loading Pass 1 note lengths into the clock, since the duration of a slice cannot be greater than that of a Pass 1 note. This having been done, the keyboard is read, and, if a new note is struck, a slice of appropriate length is stored. Once again the pass and the job can be terminated by pressing the control key. If an interrupt occurs before a new note is played, it must be because the Pass 1 note is over, and this defines the end of a slice. The slice is written to storage, the next Pass 1 note is read, and its duration is loaded into the clock. At the end of the second pass, a final quartet is written with zero duration (Word 3) as an end marker.

The quartets are structured into files stored on the DF32. These are subsequently copied to LINCtape using OS/8 file handling software, where they can be accessed for editing and from which they can be restored to the disk when required by MUSICP.

MUSICP

This program reads time slices of music from the DF32, presents them to the subject and measures his

Figure 2. Music storage in MUSICW.

- f_1 frequency for channel 1 output
- f_2 frequency for channel 2 output
- t duration of note or slice
- v control information for volume and timbre of notes

reaction times to target notes. Figure 3(b) is the flowchart for this program.

After initialization, the music file is read from the DF32. The end of job indicator is tested first (zero in Word 3 of the quartet). The time slices are then played to the subject in sequence. The duration is loaded into the clock, the frequencies of the two channels into the DACs. The timbre switch is set according to the indicator bit in Word 4, and the relays are set to the value necessary to trigger the envelope shapers.

When the indicator for a new timbre is detected, the subject's response history is checked to see whether he has responded to the previous change in timbre. If he has not, a late response is recorded for

Figure 3a. MUSICW: Flowchart.

Figure 3b. MUSICP: Flowchart.

that target. The next check is on whether the subject has responded to the current target event, and the keyboard is scanned until he presses a key. When he has responded, or if no response is required, the program loops, waiting for an interrupt.

When an interrupt occurs, the length of the time slice is augmented to the response latency if the subject has not yet responded. The next time slice is then read and played.

File Handling Programs

A number of auxiliary programs were written in BASIC for handling and editing music files. These will be briefly mentioned insofar as they assist the system of presentation of the stimulus.

MTRIM. This program eliminates very short time slices from the music file produced by the performer in MUSICW. These originate in two different ways; contact bounce in the keys and hesitant finger movements on the part of the performer. Since they arise from two different sources, these short bursts of unintended notes have different distributions of duration. MTRIM works in a very straightforward

manner by ignoring any slice shorter than a predetermined value, but a more sophisticated algorithm is probably desirable.

MUSED. This program edits the music file in respect of the volume and timbre of the notes. Figure 4 shows the bit structure of the relevant word in each quartet.

In the series of experiments already mentioned, it was required to change the timbre of notes on one channel at semirandom intervals; i.e., every n seconds $\pm t$, where the distribution of t is random. To compensate for the difference in subjective loudness between sine and square waveforms, the volume of the note was also changed in synchronization with the changes of timbre.

MRAND. This program generates a piece of pseudomusic, the melodies of which are composed of notes whose pitch is randomly selected from the set of pitches of notes comprising the melodies of a piece of music already played by the performer. The sequence of durations of these random-pitch notes are the same as those in the original piece. These pieces of "random music" were used in a control condition of the experiment to eliminate the possibility of the subject anticipating the pitch of the notes.

In addition to these programs, the file handling programs of OS/8 are available at all times during the running of the system.

CONCLUSION

The system described is capable of generating a two-channel musical stimulus with control over the pitch, duration, and volume of the notes, and of recording the latency of subjects' responses to target notes. Modification of the system would lead easily to greater generality. For example, four DACs would make it possible to compose four-part music, at the cost of shorter time slices and an extra word of storage for each slice to hold timbre and volume information. The use of a different input program in place of MUSICW would allow the manufacture of stimuli of a more general kind. Note lengths might then be specified rather than being left to the interpretive style

Figure 4. Detail of contents of Word 4 of the quarters.

of an individual artist. This would enable the production of two-channel tonal stimuli of quite a general kind, precisely controlled for duration and intertone interval.

This software can be readily modified to drive a variety of commercially available synthesizers and synthesizer modules. In the present implementation, the LINC part of the computer is used as a peripheral to drive one sense line and three relays. Alternative peripheral digital I/O can be attached to any PDP-8, eliminating the need for the LINC subsystem.

REFERENCE

KAHNEMAN, D. *Attention and effort*. New Jersey: Prentice-Hall, 1973.

NOTE

1. The value 4095 is the greatest that can be loaded into the clock and this restriction has the effect that note lengths are calculated modulo 4.095 sec. Put another way, this restriction excludes the storage of notes longer than about 4 sec, but in normal musical performance this is not found an unreasonable limitation.

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