RANDOM ACCESS EDITING OF DIGITAL AUDIO

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ABSTRACT

The most striking characteristic of digital audio, and possibly the most controversial, is its intrinsically high sonic quality. However, this may not be the most significant benefit in terms of commercial applications. Characteristics such as archivability, flexible processing techniques, time base independence and rapid accessibility offer efficient and powerful capabilities. This paper discusses the implementation of one unique feature: random access editing. Through the use of large capacity rotating magnetic media and a smoothing buffer, it is possible to create and/or modify splices rapidly, audition them, then play the various cuts in one continuous stream. Such a system also enables various forms of processing (including such standard functions as fading, mixing and equalization) to be imposed on the signal, as well as enabling different forms of interaction including display of audio waveforms.

0.0 INTRODUCTION

The first world-wide commercially available digital recording and random access editing system was publicly demonstrated at the 56th convention of the Audio Engineering Society in New York City, November 1976 [1]. Since that time, the high sonic quality of digital recordings has received significant attention. Digital audio has many other beneficial characteristics, however, that will undoubtedly help secure its place as a viable recording technology.

These unique characteristics include freedom from generation loss during copying, the ability to play repeatedly without degradation, archiving without gradual deterioration over time, the ability to perform complex digital processing (such as deconvolution [2] and mixing [3], [4]), and the fact that the time base of the recording is "frozen" thus allowing rapid and flexible operations to be implemented.

An example of the latter is random access editing, a process that has been used by Soundstream since early 1978 to edit over 200 recordings. This paper discusses the operating characteristics and design criteria for one form of random access editing and will outline a new system architecture that has been developed.

1.0 GENERAL CONCEPT OF EDITING

Editing in general is a highly skilled and important part of the creative process whether it be in writing, film or video production, or the recording of music. It allows the creator freedom in producing the desired result.

The process is essentially the same regardless of the application. A variety of source material is created, only part of which may eventually be used. The material is examined and desirable portions are selected. They may be replicated and/or processed in some way and a sequence determined. The final selections are then joined in some fashion that provides a smooth transition to produce the desired result. The process may then be repeated for refinement.

This paper is concerned primarily with the editing of recorded sound (e.g., music) although the concepts to be discussed have general application to all forms of editing. Historically, several approaches to editing music recordings have evolved. Initially (and recently in the case of direct-to-disk recordings), recording would simply be repeated until a suitable product was realized.

With the advent of magnetic tape, the concept of "cut-and-pasting" source material developed. With this technique, actual source tapes were altered making it difficult to make changes. As multi-track recording came into existence, the technique of "punch-in" and "punch-out" became popular. Again, the original sources were altered but this time magnetically and, of course, during their actual creation [5].

The development of digital recording has resulted in additional editing techniques. In one approach, digital tape is physically cut. However, to avoid an audible thump at the splice point, it is necessary to use an electronic process to provide an undetectable transition [6].

In another approach, a form of electronic editing, similar to that used in video editing, is available. In this process, selected takes of the source material are copied to a destination tape with an electronic transition being used at the splice points [7]. This avoids the need to physically alter the source material, but the order in which editing proceeds must still correspond to the final edited sequence; changing a splice once one has moved on can be time consuming.

A third approach (actually one of the first to be used in digital editing), uses random access of the digital recording. With this process, the desired sections are again copied to a destination tape. However, the sources are first copied, in any order, to a random access memory (i.e., computer disk) from which the editing is done. This eliminates the necessity of working in sequential order. As with tape-to-tape copying, processing is used to provide undetectable splice transitions and copying is used to assemble the selections into a final continuous sequence [8], [9].

The Soundstream editing system, initially developed seven years ago, uses the random access approach. However, a significant new technique has been employed within the last two years: a time base smoothing buffer is used to produce the final edited material. The smoothing buffer essentially joins discontinuous selections into a smooth, continuous result without the need to copy. Note that only discontinuities in time base of the signal are affected--signal values are not changed. We have called this a "Continuous Random Access Editing System (CRAES)." As will be seen, this allows much faster and more flexible editing.

2.0 DESCRIPTION OF THE EDITING SYSTEM

The use of computers and random access memories to process digital audio predates the commercial application of digital recording technology. The technique has been used for many years both in a research environment (e.g., [10]) as well as, more recently, in commercial recording (e.g., [8], [9]). While the application and specific design of the systems vary, all have a common feature: real-time playback of a final set of audio data requires that the digital samples be located contiguously on the computer disk. Thus, at some point in using the system, copying is required to assemble the various pieces; this, of course, takes time, reducing the efficiency and degrading the interactive nature of the system.

The basic concept of a continuous random access editing system is shown in figure 1 (note that this system is not limited to audio editing applications). From this block diagram, the following several principal elements are apparent.

First, a large random access memory (e.g., rotating magnetic disk) is used to store the various takes of the source recording. The selections are stored in any convenient order via a transferring process in which the selections are digitally copied from the original sources (for example, a portable digital tape recorder). The selection sequence in the random access memory can be arbitrary and need not correlate with the final edited order. This assemblage, then, serves as the data source during editing.

Second, a small volatile memory holds the bridges used to produce the splice point transitions. Because these bridges are small compared to the total amount of audio, this volatile memory can be two to three (or more) orders of magnitude smaller than the random access memory. The bridges are nothing more than a properly contoured cross-fade (or smear) between the lead-out of one take and the lead-in of another. As splices are created, auditioned, and accepted, these "smears" are saved in the volatile memory awaiting final playback. As will become apparent, part or all of the edited material may be auditioned at any time during the editing process.

Third, a time base smoothing buffer is used to create continuous output from the discontinuous selections in the random access and the volatile memory. The use of this buffer eliminates the time consuming need to copy the various takes and bridges into a single contiguous section. The details of the smoothing buffer are discussed later.

Fourth, a splice table (or "menu") is created during the editing process to control playback of the recorded material. This table consists of instructions that describe what portions of what selections are to be used, the order in which they are to be used, and the parameters of the cross-fades at splice points. The table can also contain instructions describing optional processing (such as fades) to be performed on particular sections.

Fifth, a digital controller is used to interact with the system. During playback (which may occur at any time during editing) the controller uses instructions from the splice table to select appropriate material from the random access and volatile memory, and to direct that material to the time base smoothing buffer. The smoothing buffer then creates the continuous program, in real-time, and presents it for listening (or for digital transfer to a suitable digital recorder). Thus playback occurs virtually instantly (without the need to copy) according to the menu created during editing.

Sixth, a console and graphics package are used to provide the means of interaction with the system. Through the console, the splice table is created by auditioning the selections, indicating splice points, specifying splice parameters (which are flexible over a wide range of values), auditioning the proposed splice, and then saving the related parameters in the menu. The graphics package may be used to visually interact with the audio to assist in refining particular splices.

Finally, optional signal processors may be imposed at various points in the system. Depending upon its nature and the specific hardware configuration, the processing may or may not be in real-time. For example, as shown in figure 1, data could be processed (in this case, in real-time) before entering the smoothing buffer to fade, equalize, etc. Processing could also take place at the output of the smoothing buffer or even in-place in the random access memory prior to playback.

Using such a system for editing, then, reduces to a simple concept: creating the splice table. Once the original source data has been transferred into the random access memory, all other elements of the editing already exist. Since the splice table serves as the master control of the editing process, modifications to the table are instantly reflected in the final result. By creating several such splice tables, all of which refer to the same source material in the random access memory, several proposed splices or even several complete jobs can be instantly auditioned and the best selected. And, of course, since all the source data can be accessed at random, there are no editing delays imposed by rewind or fast-forwarding time.

3.0 DETAILS OF THE TIME BASE SMOOTHING BUFFER

The time base smoothing buffer, which is the key to the operation of this system, is detailed in figure 2. As shown, the system is bi-directional. For simplicity, the following discussion refers to use of the buffer during playback; operation as an input device is similar, however.

The basic component of the smoothing buffer is a high speed electronic memory (FIFO) controlled by an external clock. The clock provides the time base for the system and thus controls the passage of audio samples out of the system.

The "input" side of the buffer receives audio data in discontinuous pieces (normally these come from the random access disk memory but, in fact, can come from any source). As samples enter the buffer they are stored in the next available location in the electronic memory, adjacent to the end of the previous set of data.

Simultaneously, samples are clocked out of the "output" side under control of a crystal oscillator. This ensures a fixed time base at the desired sampling rate. By selecting the proper clock frequency, this same system can be used to edit recordings made at different sampling rates. The digital samples can then be directed to a D/A converter, for auditioning, and/or directly to another digital storage device (through a suitable adapter, if necessary).

To provide proper synchronization, the buffer contains a controller that requests new data from the input side as needed. The design is such that the buffer remains as full as possible at all times. This ensures the greatest resiliancy in operation.

Two points should be mentioned. First, the size of the buffer memory determines the amount of discontinuity that can be smoothed. If the buffer is sufficiently large, the access to the individual groups of data can be relatively slow, thus widening the choices of technology employed in the rest of the editing system (i.e., the random access memory).

Second, the same buffer can be used universally with any digital format. As mentioned above, the basic clock (which can be supplied by an internal programmable clock or an external device) determines the sampling rate of the editing system. Also, a suitable adapter can be used to match the protocol of the system with that of any other digital storage device.

When used to input data, the buffer actually divides the continuous data from the digital recorder (or a separate A/D) into discrete sections. This is done because access to the random access memory is constrained to be in small groups separated by a slight delay. This procedure allows for the smooth, uninterrupted input of the digital samples.

4.0 DESIGN CRITERIA FOR A CONTINUOUS RANDOM ACCESS EDITING SYSTEM

Soundstream first began using random access editing in 1976 with experimental digital recordings; the first commercial application took place in 1978. Dozens of recordings were edited on that prototype system and provided experience from which to draw in specifying the design criteria for an improved, continuous random access system. Development of that system began in 1980 and was first used in May 1981. Some of the more important criteria were as follows.

1. The system had to be fast and highly interactive. It was (and is) our firm belief that the rapid ability to conceive of an idea, try it, and evaluate it, does much to enhance creativity in editing (or, for that matter, in any other aspect of recording).

2. Editing had to be as natural and easy as possible (within the context of electronic editing) so that users without specific technical training in computer technology could learn the system quickly and reliably.

3. Limitations in the capacity of the system had to be small or, if possible, non-existent. Parameters such as the length and contour of the cross-fade at splice points, on-line random access capacity, length of previews, etc., should not impose arbitrary constraints on the editing process.

4. The system should provide for as many different modes of interaction as possible during editing including aural, visual and tactile. This would allow a variety of approaches to locating and refining splices.

5. The system should be capable of editing as many tracks as possible within the constraints of available hardware. Specifically, it should not be limited to two-track recordings.

6. Editing should be precise, i.e., sample resolution, yet locating of desired material should not be cumbersome. The method for referencing locations should be natural and flexible (including time, assigned names, and/or sample number).

7. The system should be adaptable as needs became apparent and technology evolves.

8. The editing process should be independent of recording equipment standards (a universal editor).

9. Editing should be able to proceed in any convenient order with corrections or other changes easily made at any time.

10. Processing, not necessarily related to editing, should be possible including fading, level matching, mixing, etc.

5.0 IMPLEMENTATION OF AN ACTUAL SYSTEM

A variety of support hardware and software could be used to implement the concept of a continuous random access editing system. Following is brief description of the installation at Soundstream and its operational characteristics.

The basic Soundstream editing system uses a DEC PDP 11/60 mini-computer as the digital controller. The computer has 256k bytes of solid-state memory, a fast floating point box, and ports for terminals and graphics. Software is contained on a 28 megabyte disk (RK07). While a general purpose computer provides for easy enhancement of the editing system, specialized hardware could also have been used.

The random access memory consists of two 300 megabyte, removable media disk drives. This provides a maximum capacity of approximately 128 million audio samples or just over 21 minutes of stereo data per drive. Although the usable capacity of the system is unlimited (since the disk pack on one drive can be changed while the other drive is being accessed), this gives a total on-line capacity of over 42 minutes of stereo, 21 minutes of four-track and 10 minutes of eight-track audio. During editing, this is reduced slightly since, in addition to storing the source material, some area on each disk is reserved as scratch area and some is reserved to store the splice point cross-fades (this is the small volatile memory as shown in figure 1).

In practice, the editor seldom has to be concerned with what drive or disk pack is being accessed since a virtual mapping system has been created. The total set of available packs is considered by the software to be a single, linear memory. The beginning of the memory is considered to be sample 0. A desired selection is specified by location, with sample resolution, and the system determines where and on which pack the data is actually located. Each disk pack has a unique identifying number associated with it; when the editing requires access to a pack not currently mounted on either drive, the system will request that it be mounted. Locations can be specified either in units of time, where the first sample on the first pack is time 0, or by a record and sample number. A record is a block of 1024 samples; the notation 400;121 would mean record 400, sample 121 within that record. Also, as each selection is transferred into the memory, a name is assigned so that it can be referenced by this name as well.

The smoothing buffer was designed by Soundstream to interface directly to the DEC computer and the disk drive controller. It contains up to a megabyte of solid-state memory and is controlled by a programmable clock with sufficient resolution to provide all commonly used sampling rates including, 50 kHz, 48 kHz, 44.1 kHz, etc. The interface can accept digital data directly from a Soundstream digital recorder or, via one of several adapters that have been designed, from most of the other digital recording systems currently in use. Similarly, digital output from the interface can be directed to a D/A system for auditioning and/or to a digital recorder.

The buffer can be operated in two, four, or eight track mode. In the eight track configuration, data is transfered to or from disk at the rate of 400,000 samples (or 6.4 million bits) per second.

Control of the system is via a software program, called "DAP" (for "Digital Audio Processor"), that provides specific commands and keyboard functions to initiate the various editing processes. Approximately 60% of the software is written in FORTRAN with the remainder in assembly language. Interaction is with a "quiet" video terminal with special function keys in addition to the standard keyboard. Also used is a Tektronix 611 high-resolution display and a Bitpad graphic input device to facilitate full graphics capability.

The basic editing process is quite straightforward. Prior to beginning editing, the appropriate takes to be used are transferred from digital tape to disk. These are usually transferred in any convenient order (that may or may not be the same as the final edited order). As one disk pack becomes full, the system simply switches to the other drive and notifies the operator that a fresh pack should be placed on the first drive. This transferring process uses the smoothing buffer in input mode to divide the continuous data coming from the digital recorder into discrete chunks that can be written to disk. As transferring proceeds, a name is assigned by the editor to each selection. A complete table of names and locations is thus produced to facilitate locating the selections later.

Once transferring is complete, editing commences. Since the various transfer segments have been named, the editor can refer to selections by these names and audition them. Of course, since the audio is randomly accessible, there are no re-wind or fast-forwarding delays. The desired selection can be auditioned instantly. Since all takes are equally available, and no further copying is necessary, editing can begin with any splice and then proceed in any order. Splices that have been completed can be modified at any time.

As a possible lead-out for the splice of interest approaches, playback is stopped by pushing a button at the approximate splice point. Refinement is then accomplished by one of several methods. For example, the editor can move the lead-out in increments of arbitrary size down to a single sample. It is then possible to listen up to the lead-out and stop; begin playing from the lead-out; or do both with a slight pause in between. Playback during this process can be at full speed or, as is often helpful, at a slower speed.

Alternatively, the waveform in the vicinity of the lead-out can be displayed. Using a moving cursor on the display screen, the precise point can be selected (this technique is particularly useful where sharp attacks are desired). Figures 3 through 7 are photographs of waveforms for different types of music. Figure 3 is an envelope plot of slightly more than two minutes of music, while the others are for much shorter periods of time. In each of these figures, a vertical scale value of 32767 corresponds to a digital peak of approximately 10 volts. Using these photographs as examples, one can see how easy it would be to graphically place a splice exactly at the onset of a particular note or other musical event (such as a cannon blast from the 1812 Overture).

Another method of refinement, provided for but not currently implemented, is to use a "tape-rocking" simulator. Turning a knob simulates the rocking motion of a conventional analog recorder. Using one or more of these techniques provides flexibility in dealing with the variety of situations that are present during editing.

After the proposed lead-out has been determined, the process is repeated to find a lead-in. The parameters for the splice, such as the contour or the length of the cross-fade are modified, if desired, and a test splice is auditioned. If the splice is acceptable, the parameters are entered into the splice table ("menu"), the cross-fade is retained in the volatile memory (where it was computed) and editing moves on to the next splice. If it is not quite right, the lead-in, lead-out or splice characteristics can be changed and the splice auditioned again. Notice that the preview length is unlimited and that the splice can be heard by itself or in the context of any number of adjacent splices.

Because of the random access disk storage and use of a time base smoothing buffer, there is a great deal of flexibility in specifying the parameters of a splice. While most splices use cross-fades (or smears) 20 to 100 milliseconds long, some difficult material has required smears of 1 or more seconds. (It was a pleasant surprise for many to discover that, contrary to expectations, such long cross-fades not only work well in difficult situations but, in fact, often produce undetectable splices that can not be duplicated with analog editing techniques.) Also, the shape of the cross-fade can be varied to suit the material. For example, as the cross-fade is computed, the lead-out is gradually faded down while the lead-in is faded up. Because individual takes are somewhat uncorrelated, most require a cross-over point that is down less than 6 db to avoid a noticeable dip in volume.

Figure 8 shows an actual splice point between two sine waves of different frequency (440 Hz vs 5 kHz) and amplitude. The center of the cross-fade is at the vertical line. The smear is 0.01024 seconds long and the cross-over point is down 6 db. The smooth transition from lead-out to lead-in is apparent.

Figure 9 is a photograph of a display screen from an actual editing session and shows, in the center, a portion of a typical splice table. The numbers in the left hand column are used to reference individual splices. The locations (in units of time) are the beginning and ending points of each splice segment with the lead-out designated "LO" and the lead-in "LI." The names in the center column refer to the takes being spliced together and the right hand column lists the length of the cross-fade at the splice point (0.04096 seconds) and the cross-over value (-4 db). These parameters, of course, may be different for each splice. The total length of the side is listed under the table (in this case, just over 25 1/2 minutes). Finally, the rest of the screen displays various system parameters. At the top is a job code, the identifcation numbers of the disk packs currently on-line, and the date and time. The values at the bottom include the current splice point (SP), lead-out (LO), lead-in (LI), audition length (USE) and smear width (SMW). The dashed line is part of a moving display that indicates relative position during playback (the splice point being marked with an asterisk).

At any time during the editing, several or all of the splices can be auditioned to hear them in context. Changes to splices already completed can be easily made. Once all splices are finished, the entire edited material is auditioned and digitally transferred to a digital recording system for subsequent mastering. During this process, as with the initial transferring, the software informs the editor well in advance if a new disk pack must be mounted to ensure an uninterrupted procedure.

The system has other capabilities in addition to those usually associated with editing. Signal processing, such as multi-track mixdown, fading and deconvolution, has been programmed (although not all are real-time). Also, some unique techniques have developed. For example, a pop or tick (perhaps produced by a line transient or a chair being bumped) can be quickly located, graphically displayed and eliminated by simply interpolating the waveform for a few samples in the vicinity of the tick. As another example, certain sections of music can be repeated over and over again to quickly produce loops. This technique has often been used to create long stretches of ambience when only a few short, quiet sections are available.

Furthermore, because a general purpose computer is used as the controller, the system can adapt as additional needs are identified. One interesting concept that has been used is to verify the integrity of digital data. It is easy to transfer souce material onto disk, back to tape, and back to another disk and then to compare the data bit by bit. In doing so, preventive maintenance of the digital playback and transfer processes can be easily performed. It is standard procedure to qualify the system by such a "full loop" procedure prior to each editing session. If a single bit is wrong out of several minutes of audio data (a rare occurrence), the system is immediately diagnosed and repaired. The same process has been used to verify edited master tapes that have been archived for several years with the original sources to certify the absence of any degradation.

6.0 CONCLUSION

It can be seen, then, that digital recording promises much for the industry in addition to high sonic quality. Considerable development remains in all aspects of this technology before its full potential can be realized. However, with the variety of digital recorders, mixing consoles, and other peripherals becoming available, as well as the editing system just described, new creative tools and techniques are already finding their way into the mainstream of recording activity.

7.0 ACKNOWLEDGMENT

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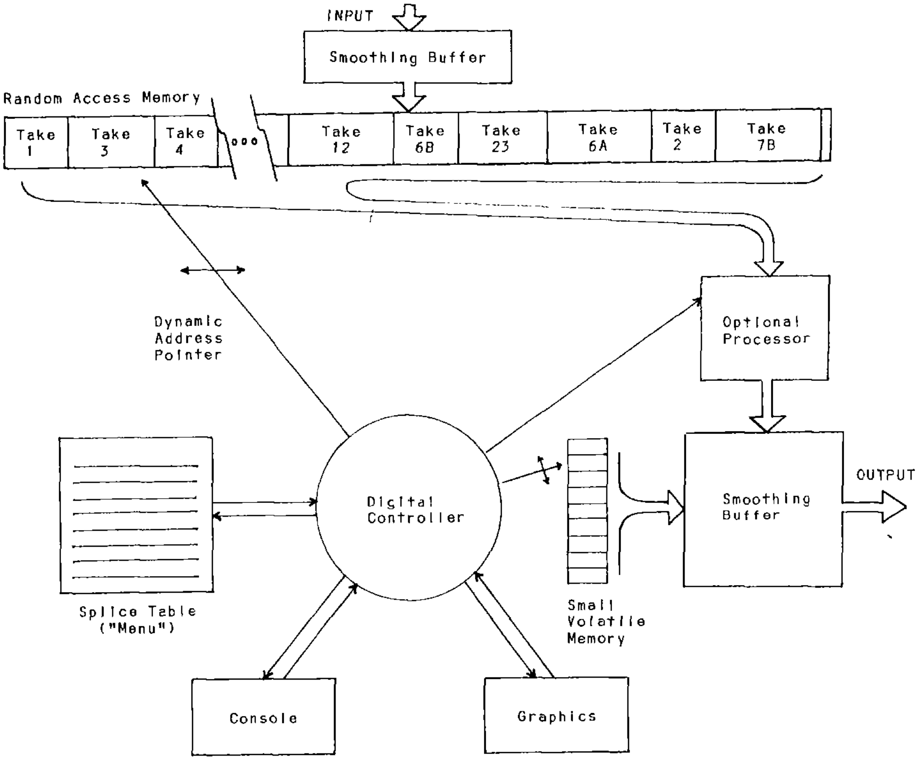


Figure 1. Block diagram of a continuous random access editing system. Note: double arrows are data paths, single arrows are control paths.

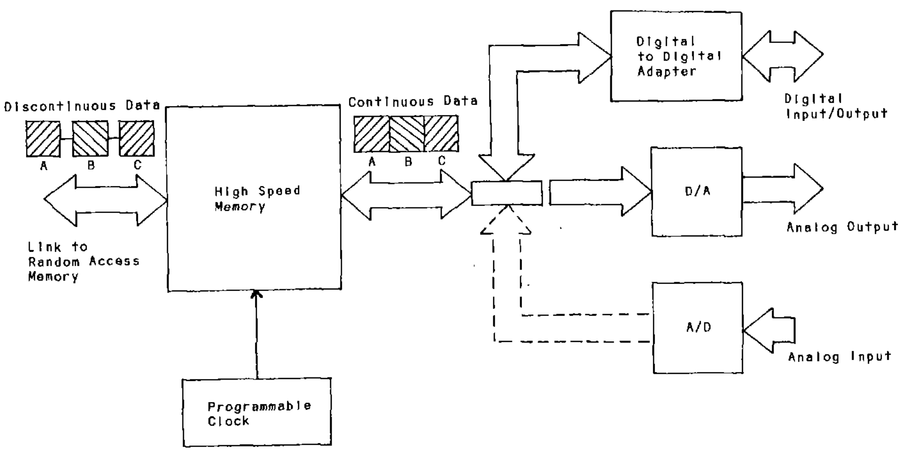
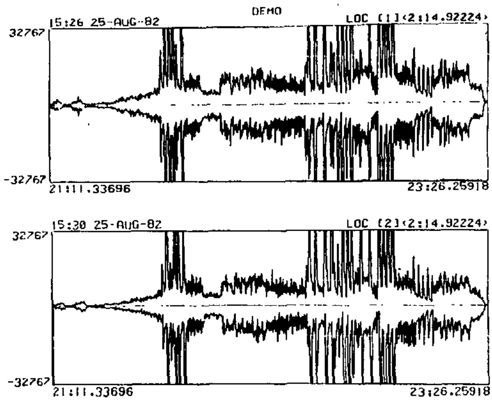
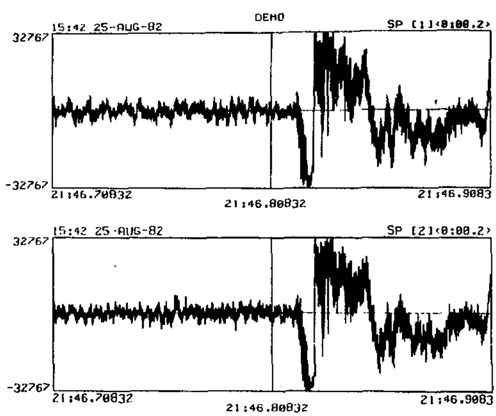


Figure 2. Architecture of a time base smoothing buffer.



1812 OVERTURE FINALE

Figure 3. Envelope display of the final 2:15 of the 1812 Overture. Large peaks are cannons. Note the extra cannons at the end of this specially edited demonstration.



1ST CANNON, 1812 OVERTURE FINALE

Figure 4. Waveform of a single cannon blast from the 1812 Overture; duration is 2 msec. The vertical scale is the full 16 bit range.

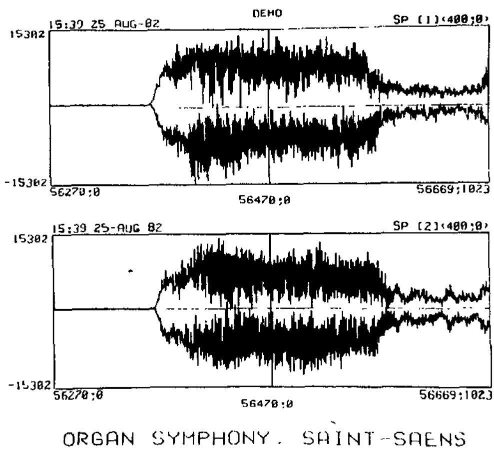


Figure 5. Organ note from the Saint-Saens Organ Symphony. Duration is 400 records (see text) or just over 8 seconds.

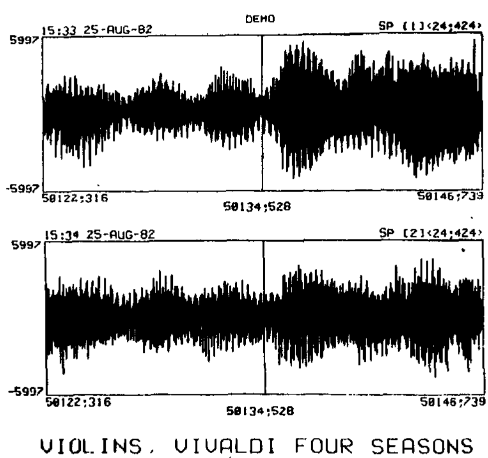


Figure 6. Violin waveforms from Vivaldi's Four Seasons; duration is 0.5 seconds.

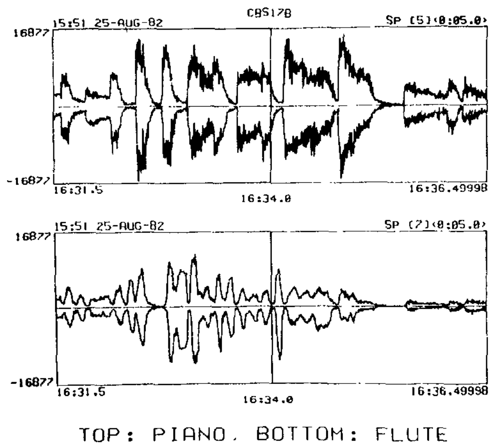


Figure 7. Waveforms of a piano and flute from an eight-track recording; duration is 0.5 seconds.

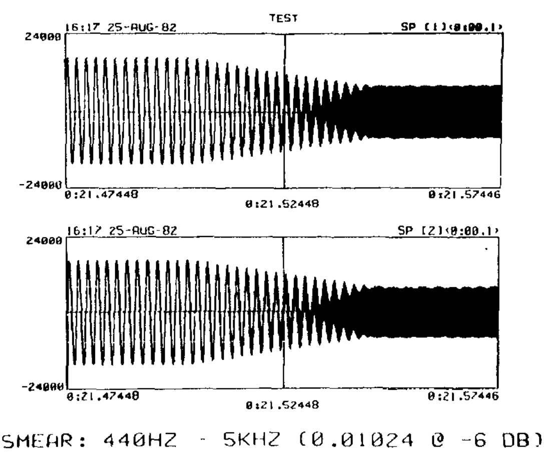


Figure 8. Splice point between a 440 Hz and a 5 kHz sine wave of different amplitudes. Duration of the cross-fade is slightly more than 1/100 of a second.

```
CBS170    Packs: 61, 78                              14:37 25-AUG-82 Wednesday
-------------------------------------------------------------------------------
            LI  (M)  0:58.418649
#  4        LO  (M)  1:52.623963   A8B:A6A;A        (0:00.021279, -4.0000 dB)

            LI  (M)  1:56.837008
#  5        LO  (M)  1:57.951569   A6A:A8C;A        (0:00.021279, -4.0000 dB)

            LI  (M)  2:03.860697
#  6        LO  (M)  2:12.126086   A8C:A6B;A        (0:00.021279, -4.0000 dB)

            LI  (M)  2:15.02081
#  7        LO  (M)  2:17.354045   A6B:A8D;A        (0:00.042559, -4.0000 dB)

Total Length   71700;700    (25:25.790545)

DAPV10>
-------------------------------------------------------------------------------
SP: (M) 17:25.379999  LO: (M) 18:04.437589 C14A       SMW: 0:00.063839
USE:   <0:45.454079>  LI: (M) 18:38.021648 C16        (8 Trks: Multi-track)

 |------------------#---------------#----+---------+--------------------------|
 ^17:23.358399   <0:04.255999>                                  17:27.614399^
```

Figure 9. Portion of a splice table from an actual editing session.