United States Patent [19]

Newell

[11] 3,789,137

[45] Jan. 29, 1974

[54] **TIME COMPRESSION OF AUDIO SIGNALS**

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[73] Assignee: **Westinghouse Electric Corporation,** Pittsburgh, Pa.

[22] Filed: **Apr. 7, 1972**

[21] Appl. No.: **241,944**

[52] **U.S. Cl.** **178/6.6,** 178/5.6

[51] **Int. Cl.** **H04n 5/76**

[58] **Field of Search**..... 178/5.6, 5.8, 5.4 CD, 6.6 A, 178/DIG. 3, 6.6 DD; 179/15.55 T, 100.2 S

[56] **References Cited**

UNITED STATES PATENTS

3,104,284	9/1963	French et al.	179/15.55 R
3,277,246	10/1966	Altonji	179/100.2 S
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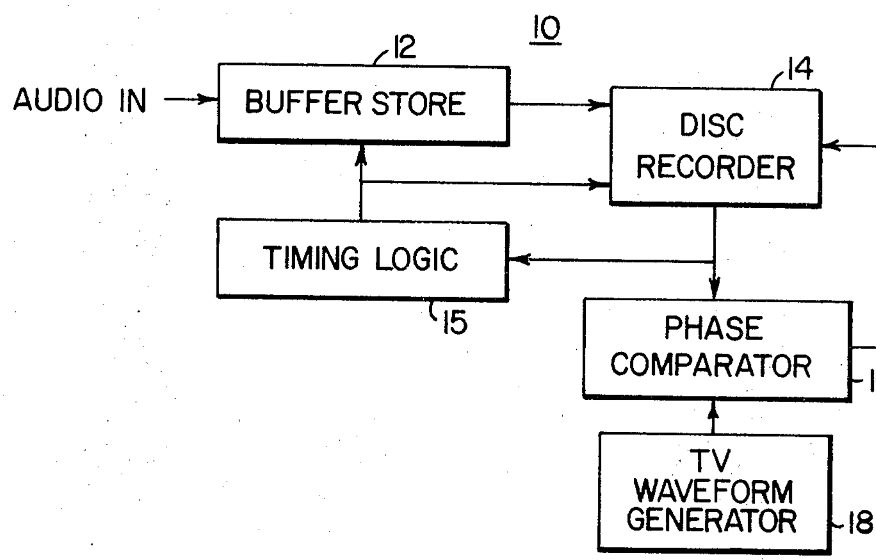
Primary Examiner—Harvey E. Springborn
Attorney, Agent, or Firm—F. H. Henson et al.

[57] **ABSTRACT**

The invention pertains to a technique for time-compressing audio signals without disturbing the relative position of the audio samples through the use of a time buffer store and a subsequent FM recording of the time-compressed audio signal in a format similar to video signals and the subsequent demodulation of the FM signals and time expansion of the signals to produce the original audio signals. The use of FM modulation and demodulation of a type similar to that used with video information for recording and transmitting time-compressed audio information permits common processing of audio and video information and the utilization of standard video transmission and video tape and video disc recorders to accommodate audio as well as video information.

14 Claims, 14 Drawing Figures



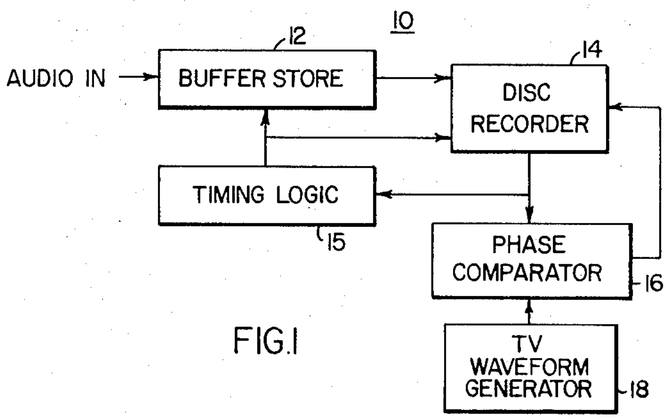


FIG.1

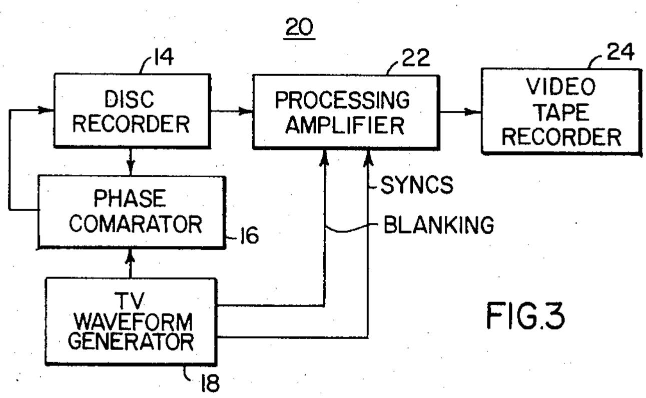


FIG.3

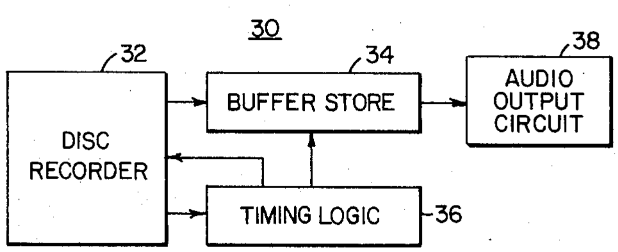


FIG.4

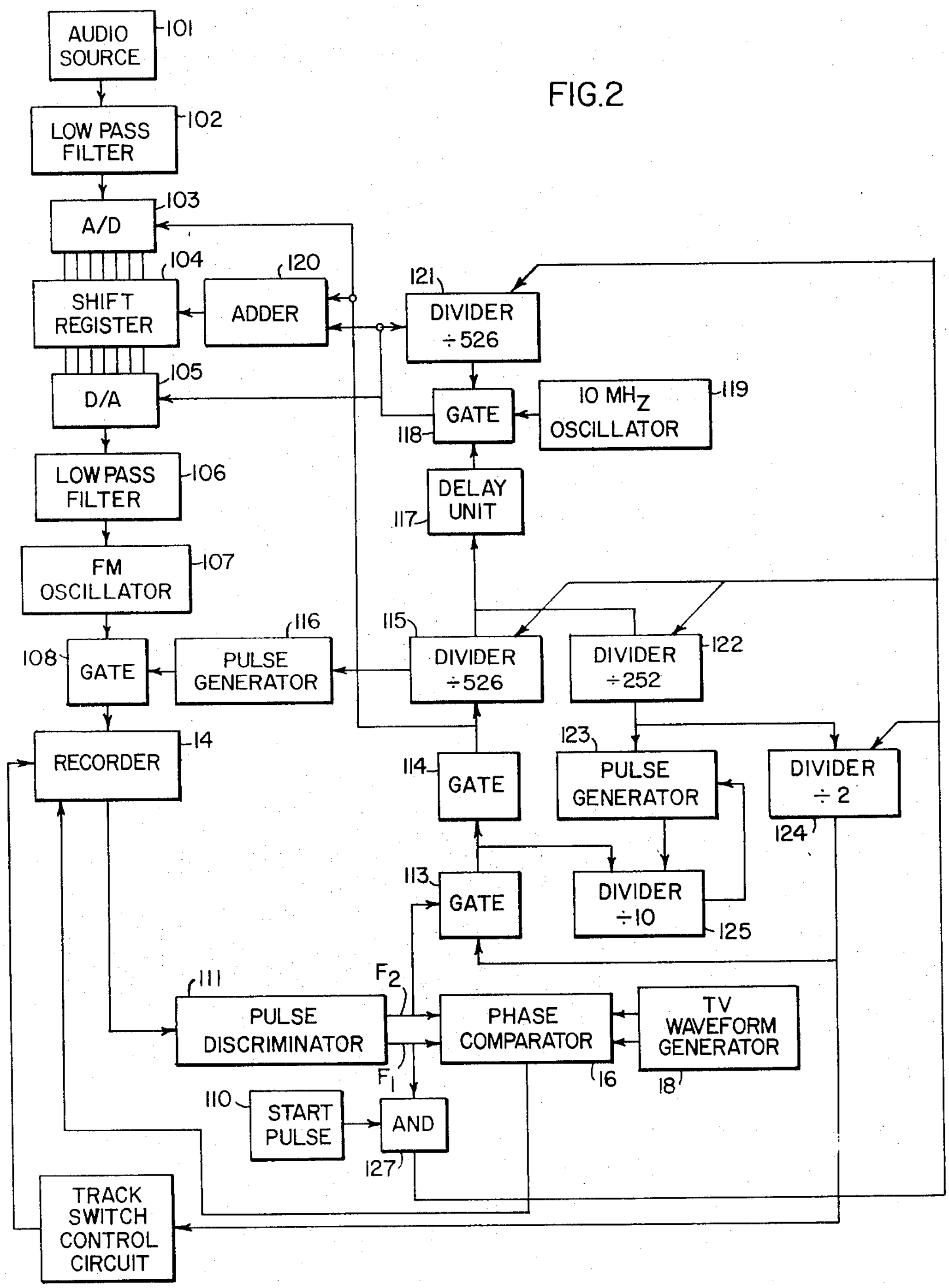

AUDIO SOURCE
101
LOW PASS FILTER
102
A/D
103
SHIFT REGISTER
104
ADDER
120
DIVIDER ÷526
121
D/A
105
GATE
118
10 MHz OSCILLATOR
119
LOW PASS FILTER
106
DELAY UNIT
117
FM OSCILLATOR
107
GATE
108
PULSE GENERATOR
116
DIVIDER ÷526
115
DIVIDER ÷252
122
RECORDER
14
GATE
114
PULSE GENERATOR
123
DIVIDER ÷2
124
GATE
113
DIVIDER ÷10
125
PULSE DISCRIMINATOR
111
F2
F1
PHASE COMPARATOR
16
TV WAVEFORM GENERATOR
18
START PULSE
110
AND
127
TRACK SWITCH CONTROL CIRCUIT

FIG. 2

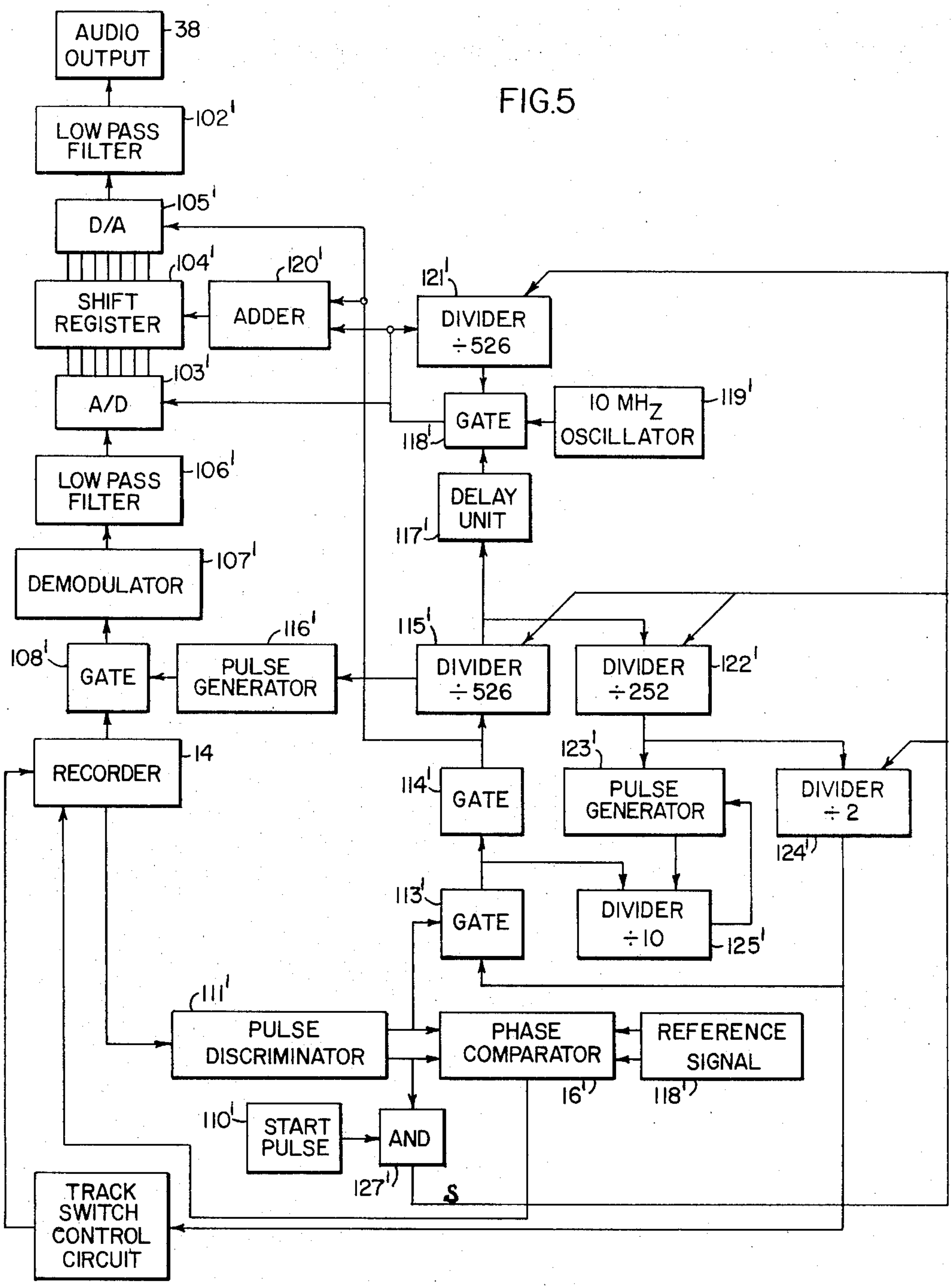

AUDIO OUTPUT
38
LOW PASS FILTER
102'
D/A
105'
104'
SHIFT REGISTER
ADDER
120'
121'
DIVIDER ÷526
103'
A/D
GATE
118'
10 MHz OSCILLATOR
119'
LOW PASS FILTER
106'
DELAY UNIT
117'
DEMODULATOR
107'
116'
115'
108'
GATE
PULSE GENERATOR
DIVIDER ÷526
DIVIDER ÷252
122'
RECORDER
14
123'
114'
GATE
PULSE GENERATOR
DIVIDER ÷2
124'
113'
GATE
DIVIDER ÷10
125'
111'
PULSE DISCRIMINATOR
PHASE COMPARATOR
REFERENCE SIGNAL
16'
118'
110'
START PULSE
AND
127'
S
TRACK SWITCH CONTROL CIRCUIT

FIG.5

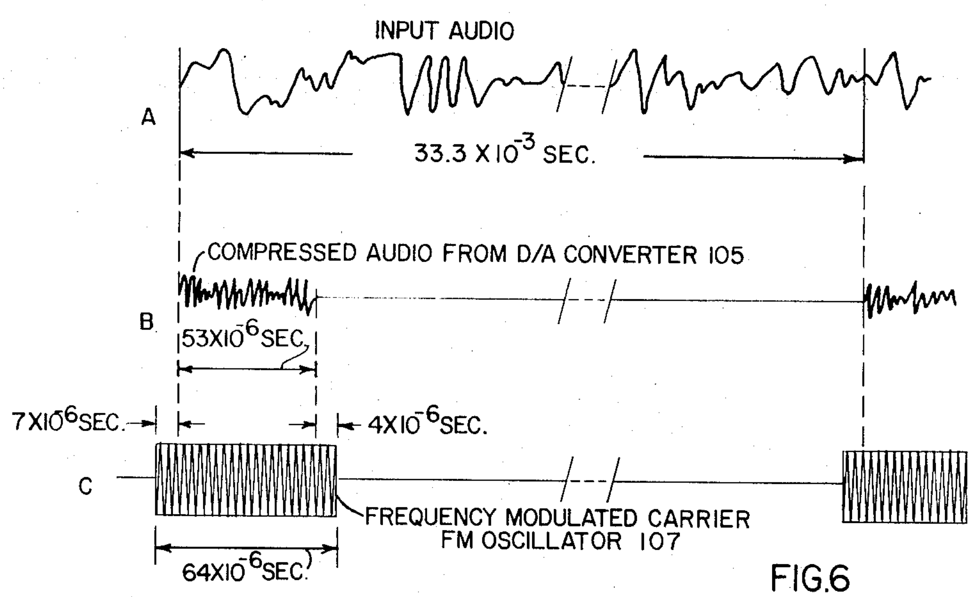

INPUT AUDIO
A
33.3 X 10^-3 SEC.
COMPRESSED AUDIO FROM D/A CONVERTER 105
B
53X10^-6 SEC.
7X10^-6 SEC.
4X10^-6 SEC.
C
FREQUENCY MODULATED CARRIER
FM OSCILLATOR 107
64X10^-6 SEC.

FIG.6

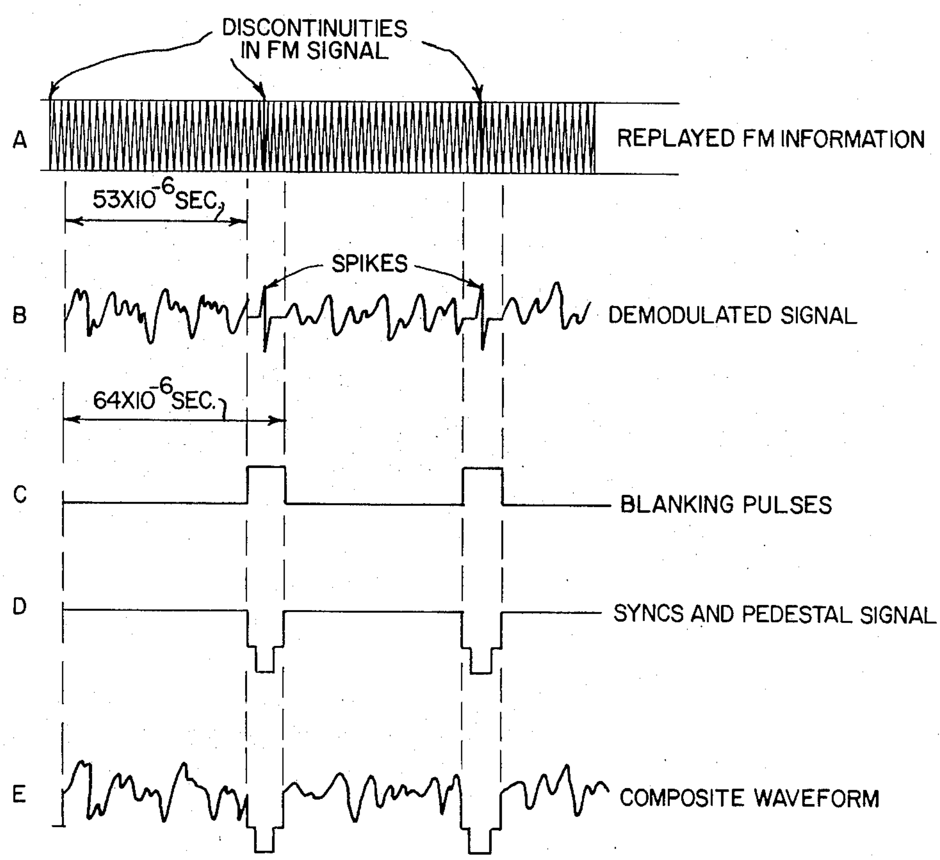

DISCONTINUITIES
IN FM SIGNAL
A
REPLAYED FM INFORMATION
53X10^-6 SEC.
SPIKES
B
DEMODULATED SIGNAL
64X10^-6 SEC.
C
BLANKING PULSES
D
SYNCS AND PEDESTAL SIGNAL
E
COMPOSITE WAVEFORM

FIG.7

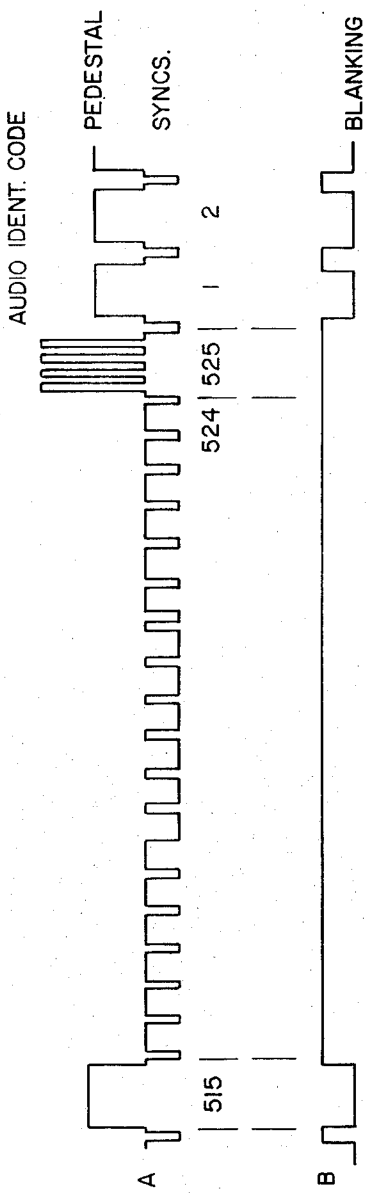

AUDIO IDENT. CODE
PEDESTAL
SYNCS.
BLANKING
2
1
525
524
515
A
B

FIG.8

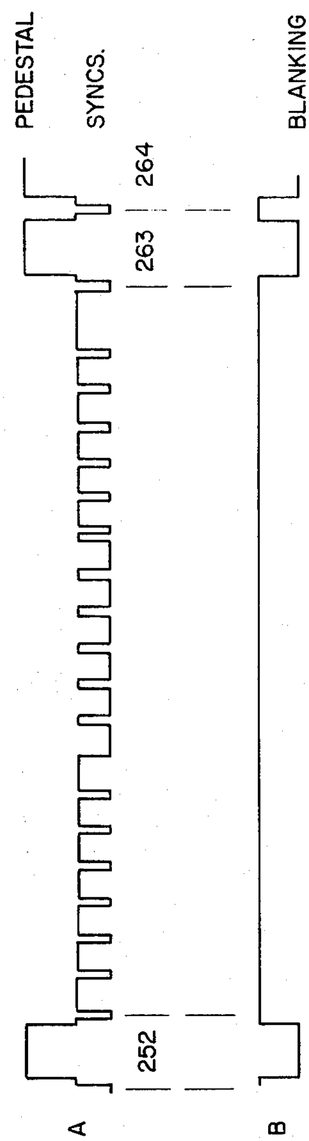

PEDESTAL
SYNCS.
BLANKING
264
263
252
A
B

FIG.9

FIG. 10

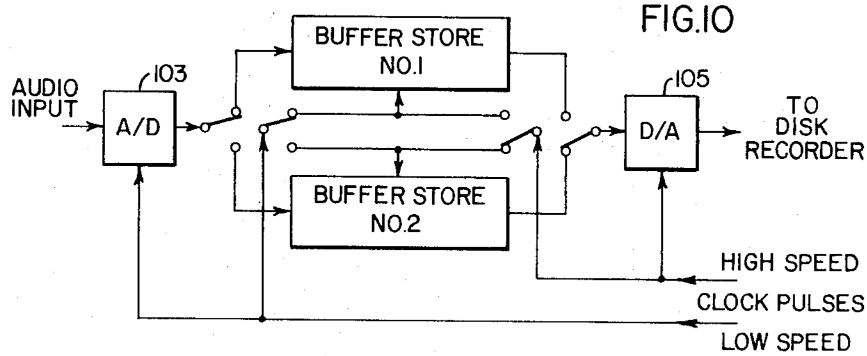


FIG. 11

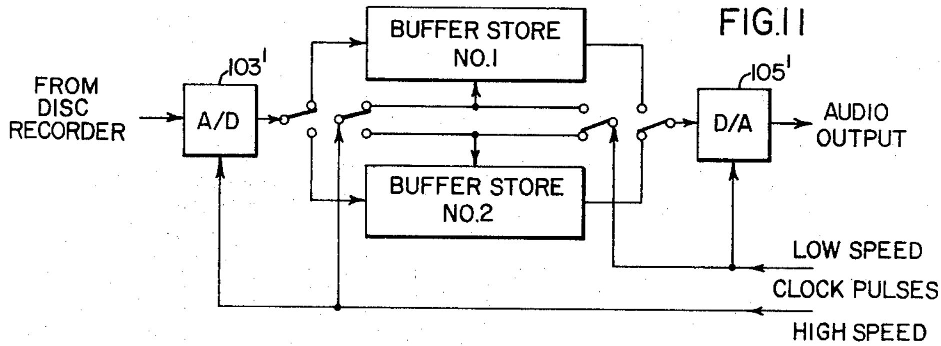


FIG. 12

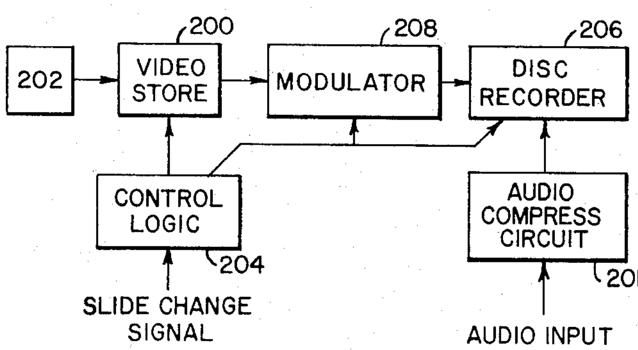
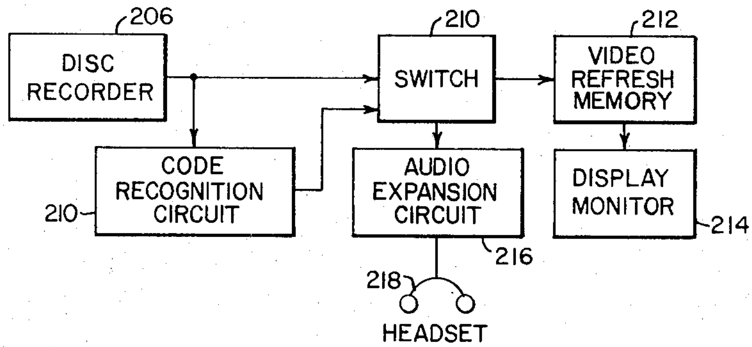
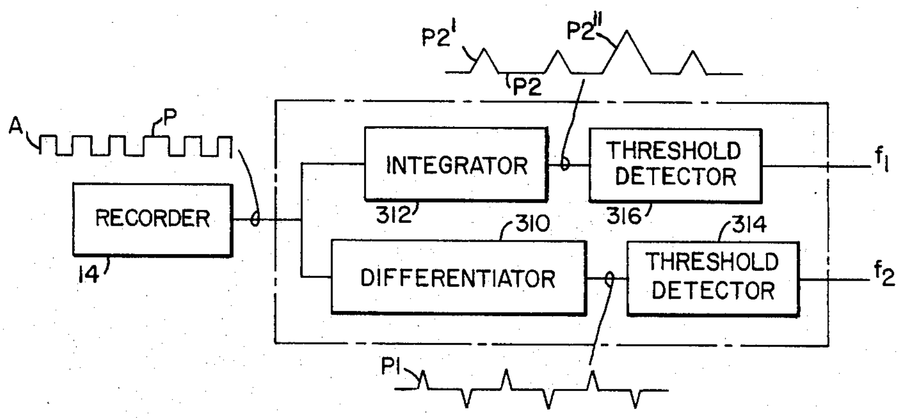


FIG.13





PHASE DISCRIMINATOR 111

FIG.14

TIME COMPRESSION OF AUDIO SIGNALS

BACKGROUND OF THE INVENTION

The time compression of audio waveforms permits the transmission of audio information via a transmission channel exhibiting an appropriately wide bandwidth in a period of time significantly short in comparison to the duration of the original audio signal. The most desirable format for time-compressed audio information is that corresponding to standard video format which would permit the transmission of audio and video signals in a time multiplex mode and permit the use of conventional TV channels and equipment for processing the time-compressed audio signals. The capability of time multiplexing audio and video signals facilitates the transmission of audio accompanied by color video stills in a time significantly shorter than the time required for presentation of the original audio information.

A previous invention disclosed in U. S. Pat. application Ser. No. 76,572 by W. W. Ramage, now abandoned, entitled "Signal Processing and Reproducing Method And Apparatus" filed Sept. 29, 1970 and assigned to the assignee of the present invention described apparatus for time compression of an audio waveform in order for record and replay by modified video apparatus. The Ramage disclosure sacrifices recording density of time-compressed audio information in that disclosure describes a technique for recording discrete samples on an individual basis. According to the Ramage technique the samples which are adjacent in the original audio signals are no longer adjacent in the time-compressed waveform and the packing density must be sufficiently low such that the intersample interference produced by the dispersion inherent in practical circuits and networks does not cause unacceptable distortion when the samples are finally rearranged in their correct relative positions. Reference is made also to U.S. Pat. No. 3,564,127 entitled "System of Band Compression for Video Signals" issued Feb. 16, 1971 to G. F. Newell and G. C. Sziklai and assigned to the assignee of the present invention.

SUMMARY OF THE INVENTION

The invention described herein permits time-compressed audio and video waveforms to be combined in time-multiplex form, recorded, replayed and transmitted over a conventional television network without modification to the network equipment. The transmitted waveforms are received and reconverted to the original audio in conjunction with the video waveforms. The disclosed invention is based on the time compression of the audio signal without disturbing the relative position of the audio samples comprising the signal. This produces a time-compressed waveform which is an accelerated replica of the original waveform except for predetermined spaces occurring at regular intervals to provide a format similar to that of video signals. The time-compressed audio thus produced is capable of being recorded by the same FM method used for conventional video and thus permits maximum recording density and therefore maximum compression ratio. Furthermore, since adjacent samples of the original audio are adjacent in the time-compressed waveform, the effects of dispersion and bandwidth limitations are no more serious than if the audio waveform were not compressed and the bandwidth and frequency limitation were scaled in proportion to the bandwidths of the time-compressed and the original signal bandwidths.

While this disclosure relates to advantages and techniques for time compression of audio, it is apparent that desirable advantages can be achieved in the combination of the time-compressed audio signals with conventional video signals. The reason for implementing the time compression of audio in a manner to convert the audio waveform into a video-like waveform with synchronizing signals added at conventional TV sweep intervals is the fact that conventional TV apparatus generally requires such synchronizing waveforms for correct operation. As an example, a video tape recorder of the type used in TV broadcasting search out the horizontal sweep synchronizing pulses and vertical sweep synchronizing pulses and uses them for the servo-control of the head-to-tape velocities and for the control of voltage-controlled delay circuits that correct for variations in the tape-to-head velocity. Conventional video control circuits and transmitter circuits include stabilizing amplifiers which use the synchronizing pulses for such purposes as clamping the voltage excursions of video waveforms. In order for an audio waveform to pass through such apparatus, the synchronizing pulses must be added and the audio information confined to those periods of the composite waveform that are conventionally occupied by the active picture information in a video waveform.

The compression of the audio waveform and the modification necessary to effect a video-like format is disclosed in the following exemplary description in connection with the accompanying drawings.

DESCRIPTION OF THE DRAWINGS

FIG. **1** is a block diagram schematic illustrating a technique for time-compressing audio signals;

FIG. **2** is a detail block diagram schematic of the embodiment of FIG. **1**;

FIG. **3** is a basic block diagram schematic of scheme for replaying in compressed form the information produced by the embodiment of FIG. **1**;

FIG. **4** is a basic block diagram schematic illustrating a scheme for replaying in the original form the time-compressed information produced by the embodiment of FIG. **1**;

FIG. **5** is a detailed schematic of the embodiment of FIG. 4;

FIG. **6** is a waveform illustration comprised of waveforms A, B and C illustrating the conversion of the normal audio input signal into a frequency modulated time-compressed audio signal;

FIG. **7** is a waveform illustration consisting of waveforms A, B, C, D and E illustrating the step-by-step technique for converting the time-compressed audio waveform into a composite waveform suitable for transmission and recording on conventional video equipment;

FIGS. **8** and **9** in their respective waveforms, A and B, illustrate the vertical blanking waveforms produced by the embodiment of FIG. **1**, including the addition of identification codes;

FIG. **10** is a basic block diagram schematic of the technique for removing gaps in the time-compressed signal during the recording of said signal;

FIG. **11** is a basic block diagram schematic of the technique for removing gaps present in the time-compressed signals during reproduction of the time-compressed signals from the recording media;

FIG. 12 is a block diagram schematic of a system for recording time-compressed audio and video information;

FIG. 13 is a block diagram schematic of a system for replay of the information recorded in the system of FIG. 12; and

FIG. 14 is a schematic illustration of a typical implementation of the pulse discrimination of FIG. 2.

DESCRIPTION OF THE PREFERRED EMBODIMENT

In order to use the FM recording process conventionally used for video signals, it is necessary for the audio signal to be time-compressed before application to the recording head. In order to convert the audio signal from the time-compressed space to that of a normal audio signal, it is necessary to time expand the time compressed. A typical manner for achieving these objectives is described herein through the use of a time buffer store and a magnetic disc recorder.

Referring to FIG. 1 it is illustrated in block diagram form an audio record system **10** comprising a buffer store **12**, a magnetic disc recorder **14**, timing logic **15**, phase comparator **16** and a TV waveform generator **18**. The timing logic **15** responds to clock pulses prerecorded on a track of the magnetic disc recorder **14** by controlling the audio sampling and buffer such that input audio is supplied to the buffer store **12** for approximately 1 rotation of the magnetic disc, i.e., 33 milliseconds, and then the timing logic causes the information in the buffer store **12** to be transmitted in a period equivalent to 1 active TV line, i.e., 53 micro seconds, for recording on the magnetic disc **14**. The cycle is repeated and the second burst of compressed audio input is recorded on the same track in a position immediately adjacent to the first burst of compressed audio input. As shown in waveform A of FIG. 6 the audio input is separated into bursts of approximately 33 milliseconds and compressed as shown in waveform B through the operation of buffer store **12** before being used to modulate the short burst of FM signal as shown in waveform C. The buffer store **12** can be implemented in numerous ways including the use of a parallel-bit digital buffer store and the sampled analog type described in BBC Engineering Monograph No. 63, August, 1966.

Since conventional U.S. TV waveforms for synchronizing purposes are harmonically related to a **525** line rate the clock pulses recorded on the magnetic disc recorder **12** can be any multiple of **525** pulses for synchronizing with the output of the TV waveform generator **18**. The phase comparator **16** serves as a servo control for maintaining the magnetic disc rotation speed at one revolution per TV frame period as determined by the output from the TV waveform generator **18**. The phase comparator circuit **16** can be implemented through numerous techniques well known in the television art and elsewhere.

The audio recording system illustrated in basic block diagram form in FIG. 1 is illustrated in detail block diagram form in FIG. 2. An audio source **101** such as a microphone-amplifier or an audio recorder, supplies audio signals to a low-pass filter **102** having a cut-off frequency typically less than 6 kilohertz. The bandwidth must be less than half the sampling rate and 5 KHz is a commonly used audio bandwidth for AM broadcast reception. The filtered analog signal is then applied to analog-to-digital converter **103** which in turn converts the signal to a multiple-bit digital signal, i.e., a 7 bit digital signal, which is supplied in parallel to a shift register **104**. The analog-to-digital converter **103** and the shift register **104** are driven by clock pulses at a rate which is the multiple of **525**, i.e., 14,750 hertz. The digital representation of the audio signals produced by the analog-to-digital converter **103** fills the shift register **104** to its capacity, which for this discussion, corresponds to 526 samples each of the 7 bits. When all the positions in the shift register **104** are filled, the clock rate for the shift register and the digital-to-analog converter **105** is established at approximately 10 megahertz by the output of crystal oscillator **119** for a period of time corresponding to **526** pulses. The clocking of the shift register at 10 megahertz for **526** pulses causes the information to be transmitted to the digital-to-analog converter **105** which in turn reconverts the digital signal to an analog signal which is time-compressed into a time period of approximately 53 microseconds. The time-compressed analog audio signal is then applied to a low-pass filter **106** having a bandwidth suitable for the removal of sampling structure while retaining audio information in time compressed form. The bandwidth should be less than 4 MH_z which corresponds to the video baseband frequency limit. The time-compressed analog audio output signal from the filter **106** is applied to frequency modulate the output of *fm* oscillator circuit **107**. The output of fm oscillator **107** is subsequently applied through gate circuit **108** to the magnetic disc recorder **14**. The fm oscillator **107** generates a waveform whose frequency is modulated in response to the amplitude of the signals provided by the low-pass filter **106**. Gate circuit **108** functions to gate output signals from the fm oscillator circuit **107** to the recorder **14** when enabled by a 64 microseconds pulse, which corresponds in duration to a television line, from the pulse generator **116**. Thus, in the time interval between two audio signal samples from the analog-to-digital converter **103**, as determined by the respective clock rates, the shift register **104** is cleared through the digital-to-analog converter **105** and is in condition to accept another **526** samples before being cleared again.

The timing reference for the recoridng process may typically be in the form of a predetermined pulse train on one track of the magnetic disc recorder **14**. The prerecorded clock pulse train may typically contain **525** pulses of which one pulse is identifiable by virtue of amplitude or width. The clock pulse train could, of course, be any multiple of **525** pulses and dividers used to obtain either 31,500 hertz or 15,750 hertz for synchronizing with the output of the TV waveform generator **18**. The clock pulse train as derived from the magnetic disc recorder is applied to pulse discrimination circuit **111** which divides the clock pulse train into two waveforms, the first being f_1 consisting of one pulse per rotation and the other being f_2 consisting of 525 pulses per disc rotation. The pulse discrimination circuit **111** can be implemented through the use of a circuit which is responsive to the width or amplitude of the singularly identifiable pulse. The function of circuit **111** is to identify the unique pulse and provide two outputs, an f_1 output which corresponds to one pulse each disc rotation, and an f_2 output which corresponds to a train of 525 pulses per disc rotation.

A typical implementation of a pulse discriminator circuit is illustrated in FIG. **14.** Assume a pulse train from the disc recorder of the type A wherein the single pulse P is distinguished from the remaining pulses as being of twice the duration. The pulse train is applied to a differentiator circuit **310** and an integrator circuit **312.** The differentiator circuit **310** responds to the leading edge of each pulse and develops an output pulse train P1 of positive pulses which is subsequently clipped by threshold detector **314** to produce the f_2 pulse train. The integrator circuit **312** responds to the pulse train from the disc recorder **14** by producing an output pulse train P2. The integration of the long duration pulse P results in a pulse P2′ exhibiting a greater amplitude than the remaining pulses P2″. Threshold detector **316** is set to respond to pulse amplitudes greater than that of the P2″ pulses thus functioning to generate an f_1 pulse in response to the P2′ pulse.

The phase comparator circuit **16** functions as a servo control for maintaining the disc rotation speed at one revolution per TV frame period as determined by the output from the TV waveform generator **18.** The single pulse f_1 is used in conjunction with a "start" pulse from start pulse circuit **110** which is applied to AND gate **127.** The simultaneous presence of a "start" pulse and an f_1 pulse at AND gate **127** results in the gating of signal S which serves to reset all the counters in the recording system **10** to establish a reference time for commencement of the recording operation. The f_2 pulses which occur at television horizontal sweep rates are supplied via gate circuit **113** and a gate circuit **114,** the latter of which is normally open, to provide clock pulses to the analog-to-digital converter **103,** and via adder circuit **120** to shift register **104.** Gate circuit **113** transmits the f_2 train of pulses to gate circuit **114** except when an inhibit pulse is supplied to it from divider circuit **124** to which it responds by deleting one pulse from the train. Gate circuit **113** can be implemented through the use of an exclusive OR circuit or through the use of an AND gate which has as one input the f_2 pulse train from circuit **111** and as a second input a waveform of, say unity amplitude from divider circuit **124** under all conditions except during the presence of an output pulse from divider circuit **124** at which time the amplitude to zero resulting in deletion of one pulse from the f_2 train transmitted to gate circuit **114.** This inhibit pulse from divider circuit **124** must coincide with one of the pulses from separator circuit **111.** Many other circuits can perform this function, even a simple transistor or tube that is caused to be biased below operation level when an inhibit output pulse is transmitted by divider circuit **124.** The divider circuits may typically consist of counters. The use of divider circuits is illustrated in reference U.S. Pat. No. 3,564,127.

Gate circuit **114** transmits the pulse train from gate circuit **113** which consists of **525** pulses per disc rotation and feeds them to divider circuit **115,** which in turn functions to divide by **526** and provide an output once every **526** pulses of f_2. The output of divider circuit **115** is divided by **252** in divider circuit **122** and causes a pulse waveform generated by pulse generator **123** to inhibit ten pulses of f_2 from passing through gate circuit **114.** The output of divider circuit **122** is also divided by 2 in divider circuit **124,** and the output of divider circuit **124** as described above is used to inhibit one pulse of f_2 from passing through gate circuit **113** once every disc rotation. Divider **115** generates an output pulse whenever **526** of the f_2 pulses have been supplied to the shift register **104** and opens gate circuit **118** via time delay circuit **117** to permit f_3 pulses at a 10 megahertz rate from oscillator **119** to be supplied to the digital-analog converter **105** and the shift register **104.**

Gate circuit **118** functions as a buffer amplifier that is normally switched off except when a pulse from the delay circuit **117** enables it to transmit the 10 MHz wave-form f_3 from oscillator circuit **119** to digital-to-analog converter **105** and adder circuit **120.** Having been enabled, the gate circuit **118** remains open until it has transmitted **526** pulses of f_3, at which time the divider circuit **121** divides by **526** and delivers a disenable pulse and closes gate circuit **118.** There are many ways such a circuit can be implemented. As one example, a bistable circuit can produce and enable voltage when set by the pulse from delay circuit **117** and then reset by the pulse from divider circuit **121.** The enable waveform and the pulses from oscillator circuit **119** can then be AND gated to produce the output for digital-to-analog converter **105** and adder circuit **120.**

Divider circuit **121** responds to the 526th pulse from the output of gate circuit **118** by closing gate **118.** The termination of f_3 pulses from the output of gate **118** completes one cycle of the operation of shift register **104.** This cycle of operation of shift register **104** is repeated for 252 cycles. In each cycle, the magnetic disc of recorder **14** rotates precisely one rotation plus a period of time corresponding to one TV line period of 64 microseconds. Each cycle ends with a compressed audio burst being recorded on the magnetic disc in positions equivalent to the active portion of the conventional TV lines of recording or that used for visual information. The conventional televison wave-form comprises a time-multiplexed and amplitude-multiplexed combination of visual information and synchronizing information. The line periods contain a portion of visual information that is often referred to as the active portion (53 microseconds). The remainder of the line period has the visual signal blanked out and replaced by synchronizing information.

At the conclusion of 252 cycles, the divider **122** generates an output which is applied to pulse generator **123,** the latter of which responds by closing the gate circuit **114** for a duration corresponding to 10 f_2 pulses. The closing of gate **114** causes the analog-to-digital converter **103** to ignore **10** samples of the audio input and causes the magnetic disc to rotate an additional **10** line period before the next audio burst is recorded. This unrecorded **10** line period is provided for the insertion of vertical blanking and synchronizing signals as required for processing on conventional TV apparatus. Following the recording of a second set of **252** lines on the first track of the magnetic disc, the divider **124** responds to the output of divider **122** by generating a track switch signal which is applied to the magnetic disc recorder **14** to transfer the recording process to a second track on the magnetic disc. The transfer of the recording process can be achieved by supplying a pulse to a stepping motor to move the recording head to another track or, if multiple fixed heads are used, to switch the recording signal from the output of gate circuit **108** to a second head on a second track. Simultaneous with the generation of a track switch signal, divider **124** causes one pulse from the pulse train f_2 to be inhibited by providing a pulse in coincidence at the input to the gate 113. The initiation of this inhibit signal by divider 124 following the completion of recording on one track will cause the gate 114 to remain closed for a duration corresponding to eleven f_2 pulses.

This recording process will result in the recording of 504 lines on each track of the magnetic disc with two gaps, one of a 10 line period and one of an 11 line period. The 505th line of audio will be recorded on the first line position of the second track.

In the operation just described, the pulse generator 116 generates the 64 microsecond pulse to open the gate circuit 108 and permit the frequency modulated carrier resulting from the operation of fm oscillator 107 to pass to the magnetic disc recorder 14 each time the shift register 104 transfers its stored information to the digital-to-analog converter 105. The time delay circuit 117 functions to establish an appropriate time delay such that the signal from the digital-to-analog converter 105 and the filter circuit 106 initiates frequency modulation of the carrier in the fm oscillator circuit 107 approximately 7 microseconds after the gate circuit 108 is opened by the output from pulse generator 116. The operation of the pulse generator 116 and the delay circuit 117 results in the recording of a signal format on the magnetic disc which consists of adjacent carrier bursts of 64 microseconds duration, each burst frequency-modulated for approximately 53 microseconds with a 7 microsecond unmodulated carrier preceding each burst and a 4 microsecond duration of unmodulated carrier at the conclusion of each burst. This format of each burst, i.e., unmodulated-modulated-unmodulated, permits the phase discontinuity occurring at the junction of the adjacent carrier bursts to be positioned in the blanked portion on the track of the magnetic disc which is to be occupied by horizontal syncs, thus providing adequate time for any transients caused by the discontinuity to decay before the modulation commences.

It is apparent that many other circuit arrangements can be devised to perform the logic functions described in reference to the functional operation of the embodiment disclosed in FIG. 2. One variation would involve the changing of the rate of magnetic disc rotation from that corresponding to a TV frame rate and establishing it as a TV field rate and utilizing two tracks for accomplishing the recording process described above. Furthermore, the rate of sampling the audio input signal can be changed to provide greater compression of audio signals having a narrower bandwidth and less compression for a audio signals of a wider bandwidth. A wider band audio signal can be recorded by sampling more frequently, and the track will be filled in a shorter time. Conversely, the track can record a narrower band audio signal by reducing the sampling rate and the bandwidth of the audio low-pass filter. The frequency expansion achieved with the arrangement described above is given by the ratio of the input supply time and the output transmission time of the shift register 104 and can be represented as:

$$(33,333 \times 10^{-3} + 64 \times 10^{-6})/53 \times 10^{-6} = 630/1$$

The time compression achieved is equal to the number of rotations of the magnetic disc for one complete track recording. In the example described above, this corresponds to 504/1.

The information thus recorded in compressed form on the magnetic disc recorder 14 can be replayed in compressed form for re-recording on a broadcast tape recorder by means of the arrangement typically illustrated in block diagram form in FIG. 3. The recorded tracks are replayed in sequence as a continuous signal with each track requiring one thirtieth of a second for playback. The output from the magnetic disc recorder 14 on the arrangement 20 of FIG. 3 is supplied to a processing amplifier 22 which inserts blanking and synchronizing signals supplied from the TV waveform generator 18 to which the magnetic disc recorder 14 is phase-locked through the phase control 16. The processing amplifier is a circuit used in many TV applications as, for example, camera control units where the camera output waveform is accepted as a time sequence of active line periods containing video information with random noise, spikes, etc., between these periods. The processing amplifier removes all content between the active line periods and adds composite synchronizing and blanking waveforms to the video information from the camera. The output is a conventional video composite waveform. A typical processing amplifier is the Model 1085 of the Ampex Corp.

The waveforms illustrating the input audio signals leading up to the recording on a magnetic disc are illustrated in waveforms A, B and C of FIG. 6. The waveforms A, B, C, D and E of FIG. 7 illustrate the processing of the signal replayed from the magnetic disc recorder and processed through the processing amplifier 22 of FIG. 3 resulting in the composite waveform as illustrated in waveform E of FIG. 7. The demodulation of the fm carrier is provided by the disc recorder A which supplies an output waveform illustrated in waveform B of FIG. 7 as an input to the processing amplifier 22. In the waveforms A and B of FIGS. 8 and 9 there is illustrated vertical blanking waveforms resulting from the replay process described with reference to FIG. 3. It will be noted that an identification code can be added at a prescribed location such as line 525. The waveform shown in FIG. 8 represents the output from the processing amplifier 22. FIG. 9 represents the same waveform at the adjacent vertical synchronizing period. The complete waveform for one TV frame includes 525 lines and two vertical intervals.

The material recorded on the video tape recorder 24 of FIG. 3 can consist of a large number of programs that have been individually assembled on the magnetic disc recorder 14 and transferred at different times. Each program can consist entirely of compressed audio or can be comprised of interspersed frames of video and audio to form the content of an audio-visual presentation. These programs can be replayed from the tape for distribution over closed circuit cable or transmitted by conventional transmitters. Individual programs can be re-recorded on separate remote disc recorders by arranging that each program be preceded by an identification code which enables each program to be recognized by conventional logic circuits.

Once the required program has been recorded on a disc recorder, or for that matter any appropriate recording device such as magnetic disc recorder 14, video tape recorder 24 or the remote recording devices referred to above, the program can be replayed in its original time duration through the use of circuit implementation which functions in a manner essentially opposite to that illustrated and described with reference to FIG. 2. A simple block diagram schematic of a typical embodiment for providing the reconstruction of the original information is illustrated as the apparatus **30** of FIG. **4** wherein the information present on the recording device **32** is played back through a buffer store arrangement **34** in response to signals from timing logic **36** producing an output signal from the buffer store **34** which is supplied to the audio output circuit **38** which presents the audio information in an uncompressed form corresponding to the audio information provided by the audio source **101** of FIG. **2**. The buffer store **34** can be of the sampled-analog form or can be implemented in a manner illustrated in FIG. **5** as comprising circuits **103'**, **104'**, and **105'**. The circuits **102'** and **106'** can be considered part of the buffer store or as necessary filters to be coupled to the input and output of the buffer store **34**.

A detailed implementation of the arrangement **30** of FIG. **4** is illustrated in block diagram form in FIG. **5**. It is noted that the units identified in FIG. **5** correspond essentially to the units illustrated and described in reference to FIG. **2** with the difference being that instead of having audio information flowing from an audio source to the magnetic disc recorder the flow is reversed and information recorded on the magnetic disc recorder in time-compressed form is returned through gate **108'**, the fm oscillator **107'**, filter circuit **106'**, analog-to-digital converter **103'**, shift register **104'**, digital-to-analog converter **105'**, filter circuit **102'**, and audio output circuit **38**. The *fm* oscillator circuit **107** of FIG. **2** is replaced with demodulator circuit **107'** in FIG. **5** and the positions of the digital-to-analog converter **103** and analog-to-digital converter **105** of FIG. **2** have been switched in the playback schematic of FIG. **5**. The low-pass filter circuit **106** is the same as that illustrated in FIG. **2** apart from the reversal of connections. The analog-to-digital converter **103'** is the same as the analog-to-digital converter **103** of FIG. **2** except for the change of position and the sampling frequency is now increased to 10 megahertz. The shift register **104'** is identical to that of FIG. **2**, but now the register is filled by 526 samples at a 10-megahertz rate and discharged through the digital-to-analog converter **105'** at the lower rate of 15,750 hertz. The low-pass filter circuit **102'** is the same as the low-pass filter **102** of FIG. **2**, but instead of accepting signals from an audio input source it supplies audio output signals to the audio output circuit **38** which may be comprised of an audio amplifier and a loudspeaker, a headset, etc. One significant difference between the embodiment of FIG. **5** and that of FIG. **2** is that the TV waveform generator **18** of FIG. **2** can now be replaced by a simple reference signal source to provide synchronization of the magnetic disc recorder **14** at a stable frequency. In fact, a 60 hertz power line supply may be utilized to synchronize the rotation speed of the magnetic disc recorder.

The control logic utilized to provide this reverse flow of information from the recorder to the audio output circuit **38** is the same as that illustrated in FIG. **2** to control the flow of information from the audio source **101** to the recorder **14**.

Assume that the magnetic disc recorder **14** has a set of tracks each filled with compressed audio information. The operation of the "start" switch circuit **110'** initiates the playback process when such signal coincides with a frame frequency signal f_1 at the AND circuit **127'**. The AND circuit **127'** responds to coincidence of these input signals by generating an S signal to reset all the counters. The magnetic disc rotates one revolution plus the duration of one TV line period, and the gate circuit **108'** opens to transmit the first compressed audio burst, which is frequency modulated on the carrier from the magnetic disc **14**, to the demodulator circuit **107'** which in turn transmits the demodulated output, which presents the compressed audio waveform to the analog-to-digital converter **103'** via the low-pass filter circuit **106'**. The shift register **104'** is filled with 526 samples at a 10 megahertz f_3 rate before the readout commences at the 15,750 hertz rate which corresponds to the f_2 signal. From this point in time the shift register **104'** will be continuously read out at a rate of 15,750 hertz and subsequently refilled by a short audio burst at a 10 megahertz rate between readout pulses **526** n and **527** n, where n is the number of disc rotations since the playback process was initiated. When **252** audio bursts have been fed into the shift register **104'**, the magnetic disc will rotate an additional ten TV line period before the **253**rd burst is supplied to the shift register **104'**. This lapse of time without without transmission of audio information will produce a gap in the audio output of approximately 640 microseconds in duration. This brief discontinuity in the audio output produced by this gap, which gap occurs approximately every 8.4 seconds, is of such a short duration as not to be discernible. Following 504 rotations of the magnetic disc, the head of the magnetic disc will commence reading out from a second track. This process will continue until the entire recorded program is completely reproduced. The completion of the recorded program can be recognized by establishing a predetermined number of tracks per program, thus requiring means for counting the tracks, or if more flexibility is required, a coded signal can be added at the end of each program to identify the conclusion of the program at which time the apparatus **40** of FIG. **5** is returned to a quiescent state. The gap in recording and reproduction that occurs every 8.4 seconds can be eliminated if desired by various techniques which are obvious to those skilled in the art. The use of a small shift register to handle the overflow, the use of a change of sampling rate during the **253**rd rotation, or the use of twin buffer stores as shown in FIGS. **10** and **11** are but a few techniques available. If the system were used for high fidelity musical appreciation programs, it is possible that the higher sampling rate necessary would cause the gaps to occur at closer intervals of time. In that case, the gaps would become more noticable and it may be desirable to take steps to remove them. The use of two buffer stores, or, more economically, the use of two shift registers between the A/D and D/A converters provides one method of achieving this. In FIGS. **10** and **11** the buffer stores refer to the shift registers only. By providing two stores in FIG. **10**, No. 1 and No. 2, the need for the high frequency unloading to occur between two low frequency samples is removed. When one store is filled, the input can be switched to the second store and the first store emptied at any time during the filling of the second store. Similarly in FIG. **11**, by alternate use of the two stores No. 1 and No. 2, when the **252**nd burst has been fed into one shift register, it can commence discharging to provide the output audio. The **253**rd can be fed into second shift register **10** lines later and before the first register has discharged. By this means, continuous audio output can be achieved in spite of the timing discrepancies that occur in reading the bursts from the magnetic disc.

If the embodiment described above is to be utilized for color visuals, it is necessary to use a color TV waveform generator in the embodiment illustrated in FIG. **2.** Furthermore, the frequencies referred to for f_1 and f_2 will be slightly changed to those conventionally used for the NTSC waveforms. It is possible to use a frequency for the high rate sampling signal f_3 which is harmonically related to the low rate f_2 and to derive the color subcarrier frequency of 3.579,545 megahertz from the signal f_3. As an example, signal f_3 can be 9.9,124,938 megahertz which is 630 times NTSC line frequency of 15,734.26 hertz.

The addition of the time-compressed audio and video signals in time-multiplex can be achieved in various ways. In simplest form, the process could consist of first determining the appropriate places in the audio presentation at which visuals should be displayed. If the visuals are arranged to change at intervals that are integrals of the time period to replay one track of compressed audio (i.e., 16 seconds), the first track of the multiple track disc recorder **14** in FIG. **2**, can be recorded with the first visual frame. The audio recording sequence can then commence from track **2** onwards until the next visual is required. At this point the next unoccupied track can be recorded with the next visual frame and so on to the end of the presentation. FIG. **12** shows a schematic arrangement for recording video information from video store **200** and time-compressed audio information from source **201.** The video store **200** comprises a source of visual frame waveforms. It can consist of a separate set of tracks on the same disc as is used for recording the complete program, or it could consist of a television slide scanner. When the slide change signal is generated by means of an actuator **202,** the control logic **204** connects the input from the video store **200** to the disc recorder **206** through a modulator **208** which is enabled by a pulse lasting for one rotation of the disc. Each visual waveform must contain an identification code in the waveform.

To display the visuals at the same time as the audio is reproduced in the terminal processor, an arrangement such as FIG. **13** can be utilized. A code recognition unit **210** will recognize when the output frame from the disc recorder **200** represents a visual. A switch circuit **210** connects the output for the duration of one rotation to a refresh memory **212** that can be one track on the same disc having its own record/replay head. The output from this track is displayed on the monitor **214** continuously until another visual replaces it. Once the visual has been transferred to the refresh memory **212,** the audio expansion and reproduction continues in circuit **216** for presentation on headset **218.** Each slide change will cause a break of 33 milliseconds in the reproduced audio, but this will not be noticeable.

I claim:

1. Apparatus for achieving time/bandwidth exchange to convert audio signals of audio bandwidth into time-compressed audio signals of video bandwidth for recording and transmission on video apparatus comprising, first means for time compressing audio input signals of an audio bandwidth into time-compressed audio signals of video bandwidth without significant loss of information content, second means operatively connected to said first means and adapted to respond to said time-compressed audio signals by dividing said time-compressed audio signals into spaced apart segments wherein each segment is a duration substantially equivalent to the active portion of a television line, and third means operatively connected to said second means for inserting television synchronization information into the spaces between said segments to produce audio information in a waveform substantially identical to a video composite waveform.

2. Apparatus as claimed in claim **1** including pedestal voltage means for superimposing said segments of time-compressed audio signals on a pedestal voltage and amplitude control means for adjusting the amplitude of said time-compressed audio signals within limits compatible with video recording and transmission apparatus.

3. Apparatus as claimed in claim **1** further including video recording means operatively connected to said third means for recording said audio information and fourth means operatively connected to said video recording means for time expanding said time-compressed audio signals into audio signals of audio bandwidth for reproduction on audio apparatus.

4. Apparatus as claimed in claim **1** further including video recording means adapted to respond to video information in a video composite waveform and to said audio information to record said audio information and said video information on said video recording means.

5. Apparatus as claimed in claim **4** further including time multiplexing means operatively connected to said video recording means to time multiplex record said audio information and said video information.

6. Apparatus as claimed in claim **4** further including means operatively connected to said video recording means for separating said audio information and said video information during playback, further including means for displaying said video information on video apparatus and means for time expanding said time-compressed audio signals to produce audio signals of audio bandwidth for reproduction on audio apparatus.

7. A method for achieving time/bandwidth exchange to convert audio signals of audio bandwidth into time-compressed audio signals of video bandwidth for recording and transmission on video apparatus including the steps of, time-compressing audio input signals of audio bandwidth into time-compressed audio signals of video bandwidth, dividing said time-compressed audio signals into spaced apart segments wherein each segment is of a duration substantially equivalent to the active portion of a television line, and inserting television synchronization information into spaces between said segments to produce audio information in a waveform substantially identical to the video composite waveform.

8. A method as claimed in claim **7** including the step of transmitting the audio information on video transmission apparatus.

9. A method as claimed in claim **7** further including the step of recording said audio information on video recording means.

10. A method as claimed in claim **9** further including the step of recording video information in combination with said audio information on a video recording means.

11. A method as claimed in claim **10** further including the step of transmitting said audio information and video information on video transmitting apparatus.

12. A method as claimed in claim 11 further including a step following the transmission of said audio and video information consisting of recording said audio and video information on a video recording means.

13. A method as claimed in claim 11 wherein said audio information and video information comprise a plurality of separate programs, and further including the step of identifying the audio information and video information comprising a program.

14. A method as claimed in claim 12 further including the steps of displaying the video information of video apparatus, time expanding said audio information into audio signals of an audio bandwidth and reproducing said audio signals of an audio bandwidth on an audio apparatus.

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